TELECOMS
Our Solutions Your Success

SS-16

VoIP-2-GSM

USER MANUAL

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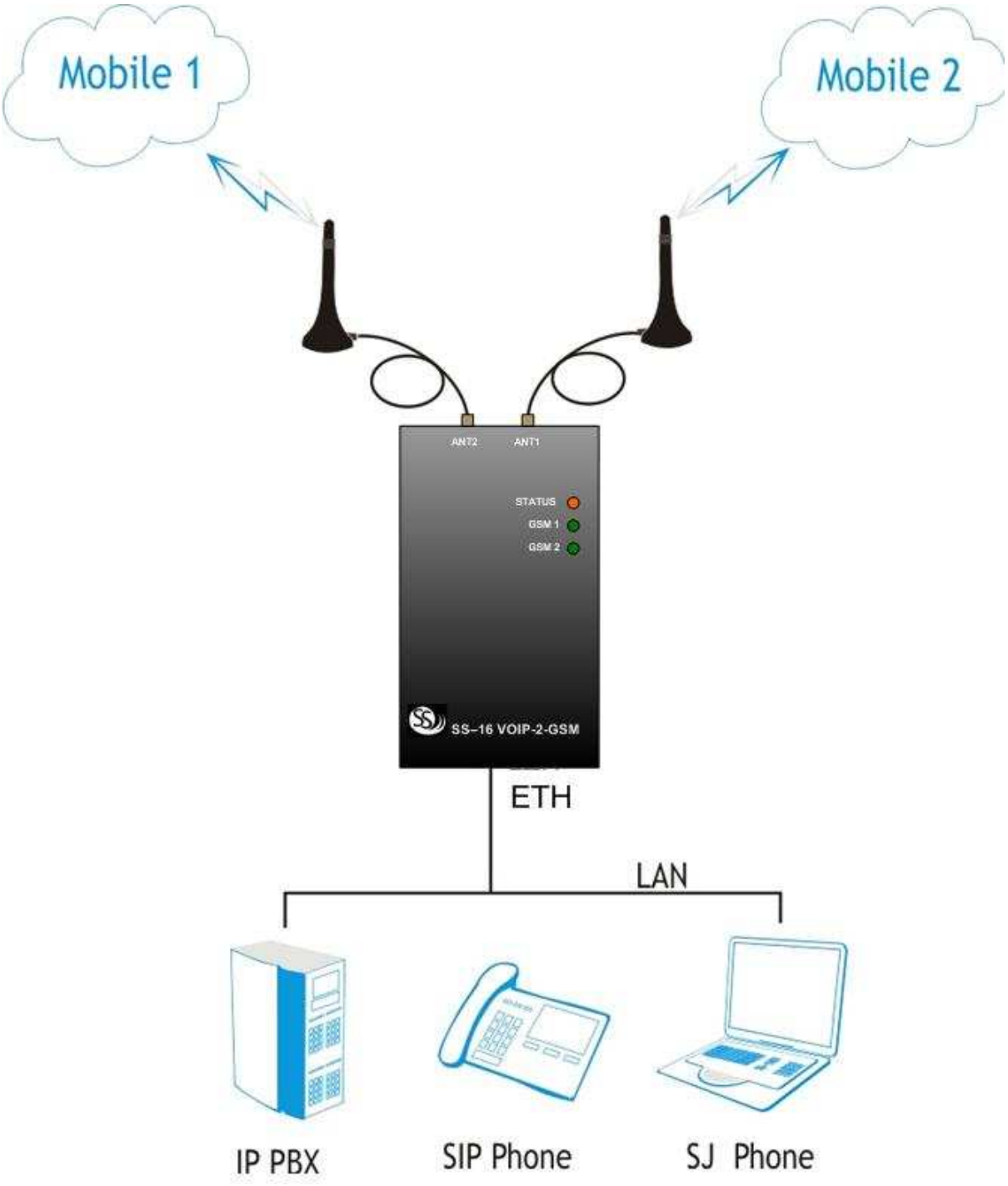
1. INTRODUCTION



REASON FOR THE INTRODUCTION OF THIS PRODUCT:

- To complete the line of SS-16 products

SS-16 VOIP-2-GSM is a new member of the SS Telecoms family of gateways. It is a compact variant of the classic VoIP to mobile gateway, featuring just two mobile modules and an ETH port. While previous SS devices connected analogue (FXO/FXS) or digital (ISDN NT/TE) phone ports to mobile networks, this one has a Voice over IP port, since it was designed to convert outgoing and incoming calls from VoIP to mobile.
The purpose remains the same, cutting down the costs of cellular calls, but the new mobile-2-VoIP interface with LCR capabilities represents a new approach in *convergence* between VoIP and home or corporate mobile communications.
By interfacing between mobile networks and IP it allows end customers to benefit by saving costs, while its two modules design provides scalability and flexibility.

- To extend the range

While the analogue or digital devices equipment allowed connections to phone terminals or PBX's located a few meters or up to 1000m away, with SS-16 VOIP-2-GSM the length of the connection becomes infinite. The mobile gateway may be located anywhere in the world and still perform an interface task.

Support, debugging and firmware upgrading can also be performed faster and from a longer distance. Previous versions of the products were connected to a local computer via serial RS-232 or USB cable. This allows management of the device, using an OAM program, sending/receiving SMS and diagnostic or update of the firmware. These operations had to be performed from the computer connected to it!

With this new SS-16 VOIP-2-GSM, it can be in anywhere in the world. The professional who performs debug or update of firmware connects to it remotely, via IP networks.

Intended use

SS-16 VOIP-2-GSM was intended to provide cost-effective access to mobile networks for the SIP users registered to the SS Telecoms IP-PBX.

Using the SS-16 VOIP-2-GSM can be beneficial for home users too, but it was intended for **office use**, both for companies that have many foreign affiliates and for small to medium companies that already own an IP infrastructure. Companies with international offices can use one or several SS-16 VOIP-2-GSM devices at each location - the calls will be routed as VoIP via Internet to the respective locations where they will be **terminated** as local GSM calls to the respective mobile networks, with minimum costs.

Enterprises with existing IP infrastructure will use the SS-16 devices to **terminate** the calls to mobile networks, achieving substantial savings on outgoing and incoming calls from IP to GSM networks and vice versa.

It can also be used by providers, ensuring low-cost call termination at different locations throughout the world. VoIP calls originating from anywhere can be sent via Internet to be **terminated** through SS-16 VOIP-2-GSM devices located at the corresponding sites. The respective SS-16 VOIP-2-GSM devices will then directs the calls to the required local GSM network, with the minimum possible cost.

Flexible Usage

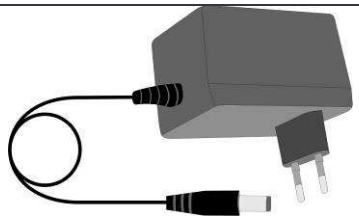
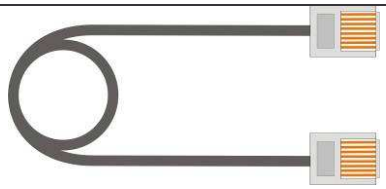
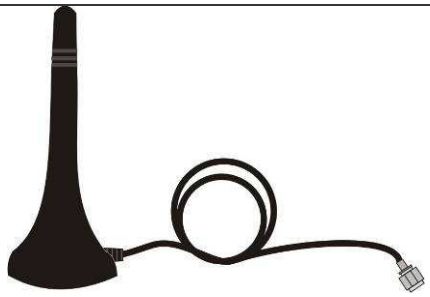
SS-16 VOIP-2-GSM can be used in several ways:

- with an IP PBX that has SIP subscribers;
- with hardware SIP telephones,
- with soft phones, such as the free application SJ Phone.

In order to ensure correct installation/configuration and good operation of the SS-16 VOIP-2-GSM device, the manufacturer strongly recommends the study of this manual before attempting operation.

2. PACKAGE CONTENT

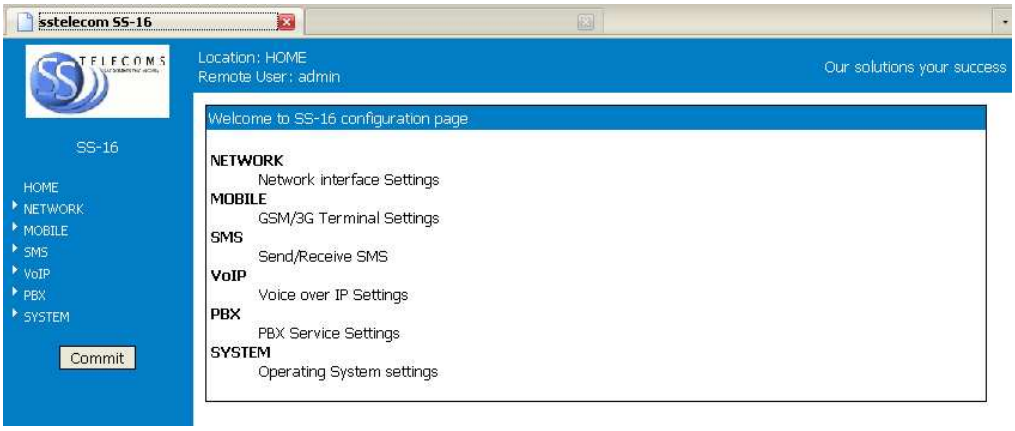
When opening the package, please check the contents against the following:

Component Image	Component Description
	SS-16 VOIP-2-GSM unit.
	Power supply: switching mains adapter Input: 100-240V $_{A.C}$ Output: 12V$_{D.C}$. / 2A Max. Power: 25 W
	Ethernet cable for local network connection A short piece of UTP straight cable, with RJ-45 connectors.
	Antennas for mobile networks Multiband, with magnetic base and 2,5 m long cable
-	CD with User's Manual

3. SS-16 VOIP-2-GSM FEATURES

The SS-16 VOIP-2-GSM device is a versatile VoIP-2-mobile gateway that cuts down interconnection costs and performs convergence between mobile voice networks and SIP calls.

SS-16 VOIP-2-GSM is fully configurable via Web pages; *it does not require installation of drivers or any additional software* on a PC. The graphics interface that shows up in the browser allows easy access to all its settings:



Since the product is Linux based, applications to enhance SS-16 VOIP-2-GSM or to customize it according to the special needs of various clients can be quickly designed by our software developers.

If signing up for our technical support offer, our best technical experts are available for technical queries. In addition, the software upgrades can be done remotely via the Internet, and are free of charge.

SS-16 VOIP-2-GSM is a highly versatile solution, when the customers require special solutions; its embedded firmware can be easily upgraded over the Internet.

4. IDENTIFICATION

4.1 LABELS AND IDENTIFICATORS

On the bottom of the case of the SS Telecoms device there are several labels or tags that indicate the characteristics and compliance.

4.2 SIGNIFICANCE OF LABELS

These adhesive labels contain certification, approval and compliance information of ICASA.

The label may include barcodes and refer to:

Network identification

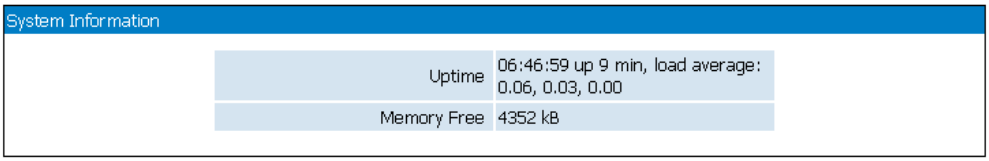
- IMEI code, International Mobile Equipment Identity for SIM based equipments. A unique 15- or 17-digit number such as <<269751923786501>> that identifies an individual mobile station to a GSM or UMTS network handset. The IMEI code is on all GSM and UMTS mobile terminals, commonly found in Europe, Asia, Africa and increasingly in America. If the SS Telecoms device features several mobile modules – for instance SS-16 VOIP-2-GSM has two modules - it will have, correspondingly, more IMEI labels - one for each mobile modem;

4.3 SOFTWARE INFORMATION

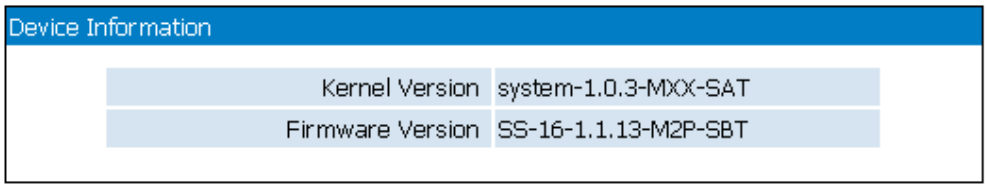
The Web interface displays some of the information described above, but also important additional information, which is NOT available through the adhesive tags. Such information is related to the serial of the motherboard of the device, or the software version actually running on it.

Two categories of information are displayed:

- System info, with temporary (current) data, such as uptime, load and Ethernet link status.



- Device info, permanent, which is important for this chapter – Firmware and kernel versions.



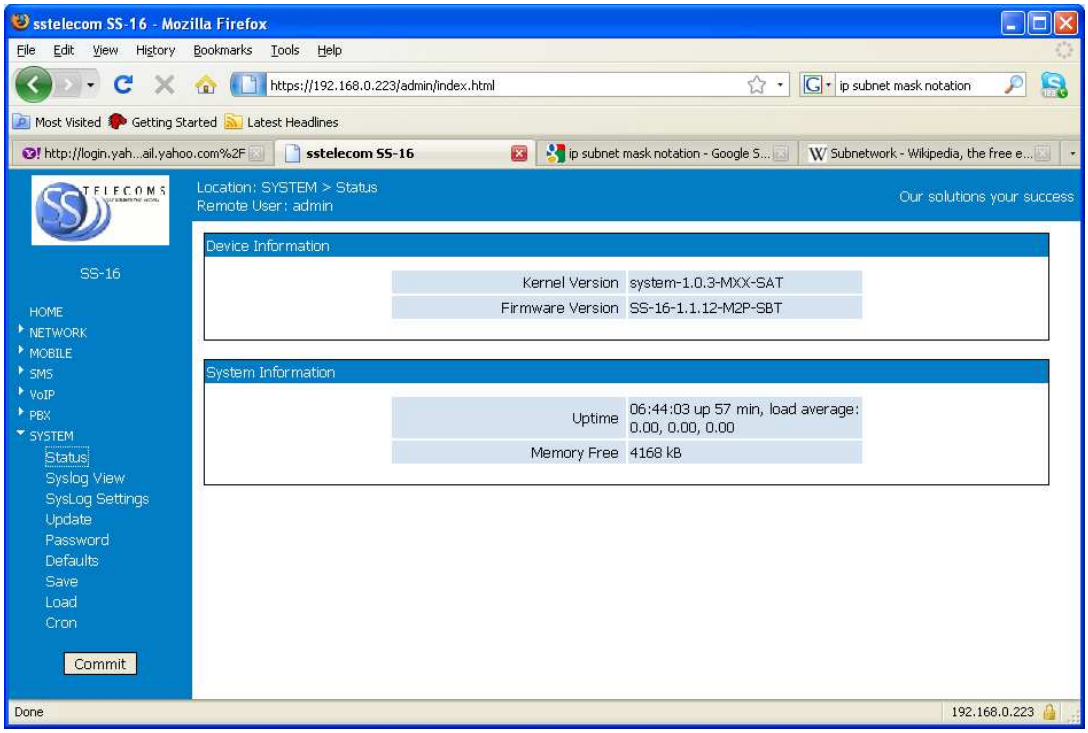
The permanent identification info shown refers to the following elements:

- Firmware version: version of the application firmware running on SS-16 VOIP-2-GSM: you may see “Firmware version is **SS-16-1.1.13-M2P-SBT**. The firmware can be easily updated (a new version) or upgraded (new software features added);
- Kernel version (Operating system version), such as **system-1.0.3-MXX-SAT** in the example above. The application firmware and the operating system (root or kernel) may be updated separately, according to the requirements of the users. The operating system memory is write-protected, thus in case of problems, accidental erasing or power failure during updating, there is still a functional “kernel” allowing loading of a new software image;

System Status

The information about current firmware and kernel can be found in the option Status – *System Information* of the menu page SYSTEM, as shown next.

For this you must access the System> Status page. Just enter into the browser the URL for the page: https://192.168.0.225/



In order to see this, one must be logged-on to the SS-16 VOIP-2-GSM device.

5. INSTALLATION

In order to ensure the proper operation of the SS-16 VOIP-2-GSM equipment, please follow the set-up steps shown below:

- Determine the exact type and model of SS Telecoms equipment
- Establish the best location
- Mounting (Hardware installation)
- Identification of connectors
- Connecting the Ethernet cable
- Disable PIN code request for the SIM to be used
- Configuring and installing the SIM cards
- Connecting the external antennas for the mobile networks
- Power up

5.1 ESTABLISHING THE BEST LOCATION

In order to determine the best location for the SS TELECOMS mobile gateway, consider the following:

- The length of the Ethernet cables that connect the SS-16 VOIP-2-GSM gateway to the PC or the network switch must not exceed 100 meters.
- SS-16 VOIP-2-GSM should be placed on a flat, sturdy surface located as far from the ground as possible. A high location, on the wall or on top of a desk or a shelf is best for the mobile connections, be it with GPRS, UMTS or GSM networks. Also, SS-16 VOIP-2-GSM should be kept clear of obstructions and away from heat sources, direct sunlight and heavy-duty electrical equipment.
- To ensure good coverage of the entire wireless mobile network in the area, SS-16 VOIP-2-GSM should be installed in a central place in the building. Normally the antennas should be in a vertical position, but if reflections occur, better results may be achieved by changing the orientation.

- For power supply, use only the adapter shipped with SS-16 VOIP-2-GSM. The jacks of different power supplies may fit but the polarity, current, voltage or regulation factor may not be compatible.
- While the device is in operation, the antennas of the SS-16 VOIP-2-GSM unit should be at least 30 centimetres away from any human being.

5.2 MOUNTING (HARDWARE INSTALLATION)

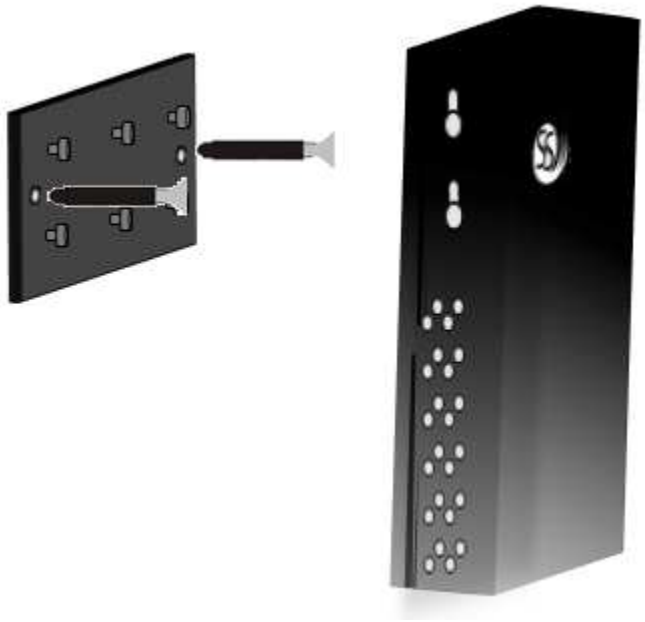


Figure 1. Mounting SS-16 VOIP-2-GSM on a wall

The SS Telecoms SS-16 VOIP-2-GSM router may be mounted either horizontally, by means of simply placing it on a flat surface, or vertically on a wall (Using a mounting bracket Available from SS Telecoms).

Horizontal mounting

The surface must be level and strong enough to hold the weight of SS-16 VOIP-2-GSM together with all its cables (power supply and wired LAN connection). A desk, table or shelf is good place for the installation of SS-16 VOIP-2-GSM.

Wall mounting (Recommended)

The equipment may also be mounted on a wall. On the left side, there is a pair of mounting holes to secure the SS-16 VOIP-2-GSM to the wall.
The mobile antenna must be in an upright vertical position.

Installing SS-16 VOIP-2-GSM on a shelf higher up, with no obstructions around, ensures the best performance embedded modems.

Take care to ensure adequate cooling of the SS-16 VOIP-2-GSM terminal.

5.3 IDENTIFICATION OF CONNECTORS

SS Telecoms SS-16 VOIP-2-GSM features only two external connectors, as described below. On the bottom panel of the equipment they are, from left to right:

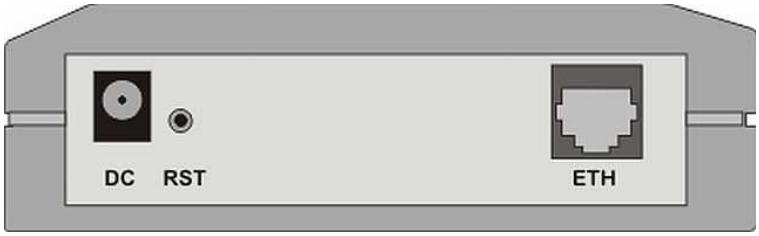


Figure 1: Connectors of SS-16 VOIP-2-GSM

- Towards the left edge, the round connector is for the power supply jack, (labelled **DC)** ($12V_{DC}$)
- The black Reset button, recessed (**labelled RST**)
- To the right, one female RJ45 connector for the local network connections **(labelled ETH**

5.4 CONNECTING THE DATA CABLE

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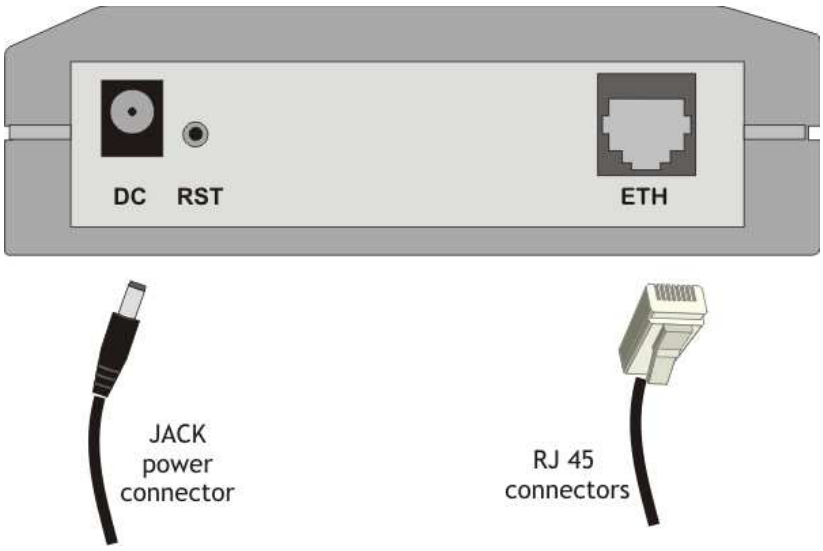


Figure 2: Connecting the cables

For Ethernet network connection:

Use standard UTP network cable (CUT 5) fitted with RJ45 connectors at both ends. The cable can be either straight-through or crossover, since the SS-16 VOIP-2-GSM is auto-crossover.
One short length of RJ-45 cable is supplied with the SS TELECOMS SS-16 VOIP-2-GSM interface.
Use either a 10Base-T or a 100Base-T connection.

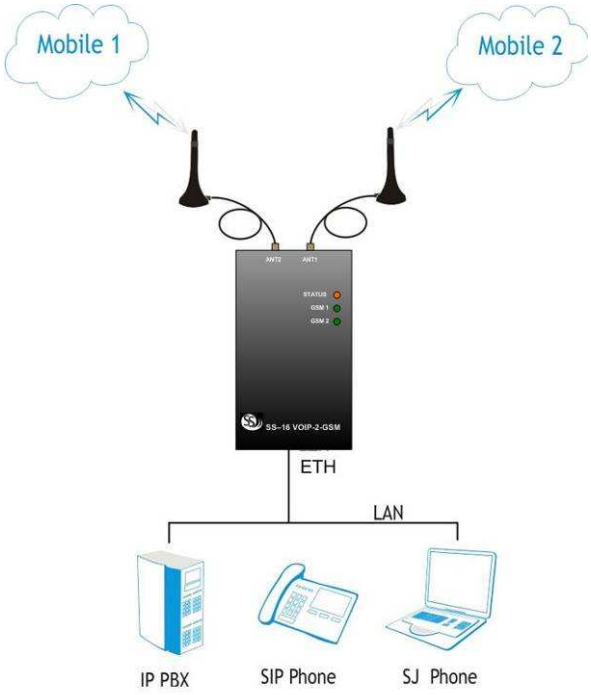


Figure 3: Connecting the LAN cable (UTP- CAT5)

The network cable is to be inserted with one end into the RJ-45 socket of the SS-16 VOIP-2-GSM device labelled **ETH**. The opposite end of the UTP cable can be inserted directly into a PC or in switches or hubs. The STATUS indicator LED of the SS-16 case will light up showing physical connection and data traffic on the respective Ethernet interface.

Power supply:
To power the SS-16 VOIP-2-GSM unit, insert the jack of the power supply adapter into the supply connector. Do **not** yet plug the adapter into the $230V_{A.C.}$ mains outlet on the wall. The special adapter from SS Telecoms, which is part of the SS-16 VOIP-2-GSM package, supplies the voltage required to power the equipment. It is an external power supply adapter.

Note: *The adapter is the disconnection device (there is no POWER switch), so the 230 V_{AC} socket-outlet shall be installed near the equipment and shall be easily accessible.*

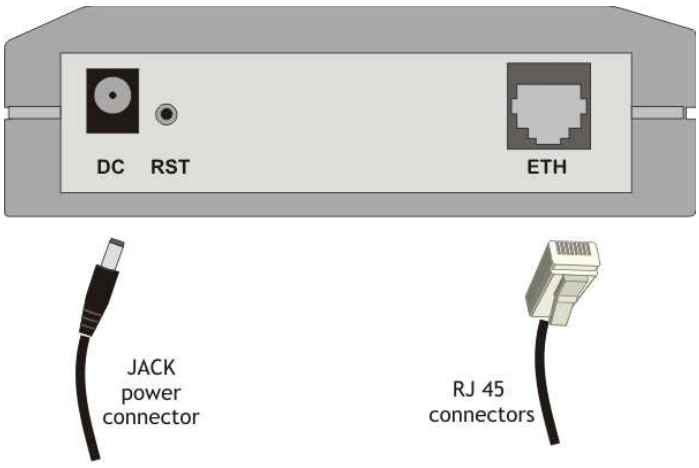


Figure 4: Connecting the power supply

Warning!

- Use only the power supply adapter shipped in the equipment package. Using other kinds of power supplies may cause damage to the equipment.
- To avoid accidents or damage to the equipment, follow the steps described earlier. First, connect the antennas, and then the power supply adapter.
- Avoid connecting or removing the antennas while the SS-16 VOIP-2-GSM device is powered.

5.5 CONFIGURING AND INSTALLING THE SIM CARDS

The SS-16 VOIP-2-GSM device may feature a single slot or two slots (in dual SIM version) for SIM cards. In order for the SS-16 VOIP-2-GSM gateway to work, it must have at least one valid SIM card with subscription to the GSM or 3G voice carrier where connection is needed.

Configuring the SIM card(s)

The SIM card(s) that is used must be active.

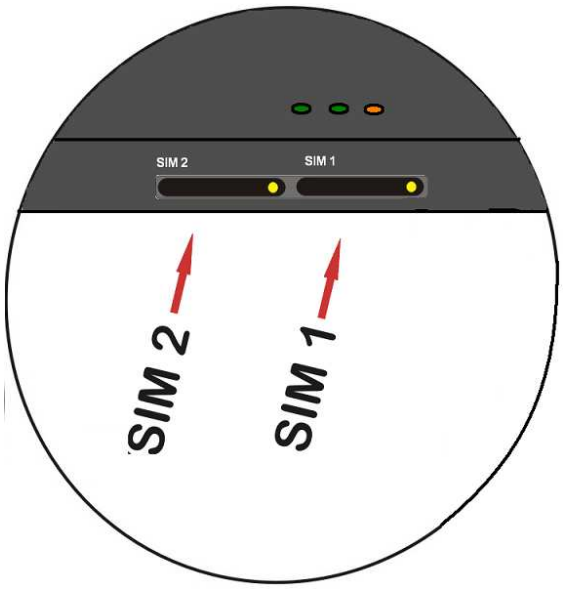
- The SIM card(s) must be configured **before** it is inserted into the slot of the SS-16 VOIP-2-GSM .
- An ordinary GSM cell phone may be used to configure the SIM card

The required configurations are:

- *PIN CODE REQUEST* – if disabling (from the menu of the mobile phone) the *PIN CODE REQUEST* security option, one will not be asked to enter it. Alternatively, the PIN code can be enabled and one may enter it form the Web page used for configuration. *In this case, take care to enter the correct PIN code **before** inserting the SIM card, to avoid PUK locking!*
- *Disable GSM services* – GSM operator offers different supplementary services for calls. When using the SS-16 VOIP-2-GSM interface it is recommended that these options be disabled because they are only available with additional costs.

Inserting the SIM cards

The one or two SIM cards to be used must be inserted into the special tray slots of the SS TELECOMS SS-16 VOIP-2-GSM device.



The slots for SIM cards are located on the side.

For inserting or extracting the SIM card use the mobile holder (removable tray) for the respective SIM card.

Use the little yellow button to actuate the holder

Handle with care when inserting or extracting SIM card.

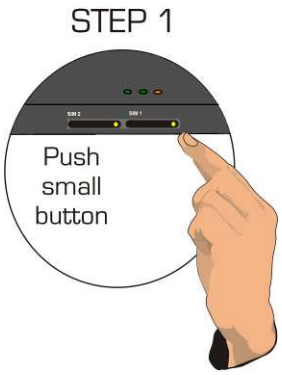
IMPORTANT:

When inserting or changing the SIM card, the equipment must be powered off.

For inserting the SIM card follow the these steps:

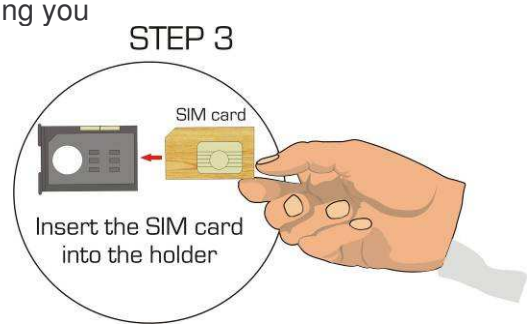
WARNING! Unplug the SS-16 unit from the main outlet before insert or replace a SIM card!
For each of the SIM cards, follow these steps:

Press the little button to eject the SIM holder.

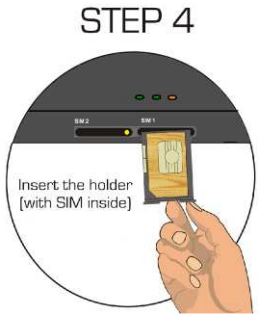


Pull out (extract) the tray (SIM holder)

Insert SIM card into the holder, as shown – with cut corner upwards and with contacts facing you



Push the holder tray with the SIM inside back into the corresponding slot of the SS-16 unit

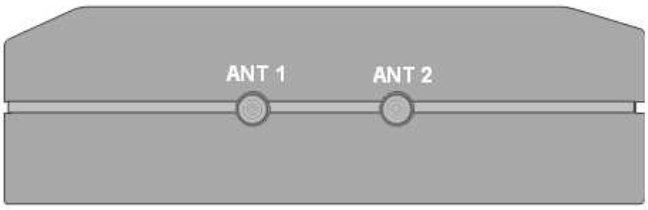


Also follow the steps described above when replacing the SIM cards already installed into SS-16.

Repeat this procedure for the second SIM and take care to insert the SIM card correctly.

5.6 CONNECTING THE EXTERNAL ANTENNAS

To ensure a good quality of transmission and to reduce radio interference use the antennas shipped in the SS-16 VOIP-2-GSM package. These antennas were designed for the respective frequency bands (multiband to allow proper connection to different mobile networks type GSM or 3G).
The antennas must be connected to SS-16 VOIP-2-GSM via the respective RF circular connectors on the top panel of the case, labelled “ANT.1” and respectively “ANT.2”.



These correspond to networks Mobile 1 and Mobile 2. The cable for each Mobile antenna must be threaded into the circular connector of SS-16 VOIP-2-GSM labelled “ANT1”, respectively “ANT2”, as shown in the following drawing.

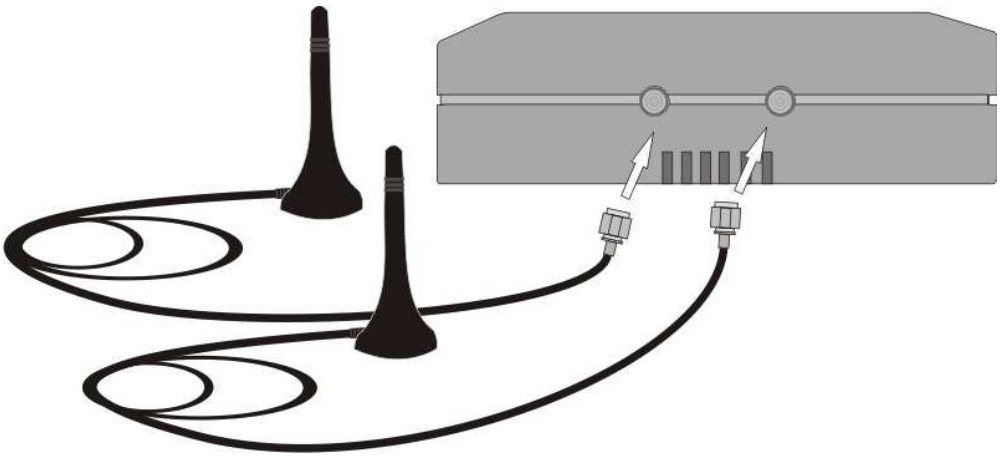


Figure 5: Attachment of antennas

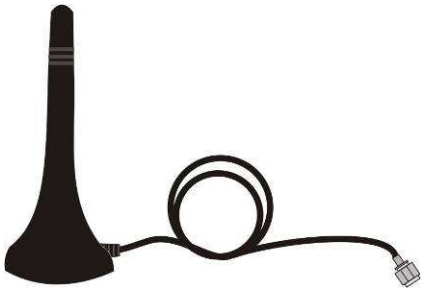
If the site has a low cellular signal, it may not be possible to use the full mobile technology without a special, high gain antenna. Such an antenna should be obtained

and installed in a higher position (the roof of the building), directed towards the base station of the mobile network carrier, in order to get best results.

Warning! *Don't use excessive force. Make sure the antenna is securely screwed into the RF connectors, but do NOT use a spanner or screw key, which could damage the antenna connector! Tighten the flange lightly, by hand.*

The antennas have vertical polarization, so they should be placed in vertical or horizontal position, depending of the local field condition.

! Place the antennas on a safe (stable) non-magnetic and flat area. The heavy base of the antenna ensures that it is secured.



The following table shows the main characteristics of the multi-band stick antennas with magnetic base (currently supplied with the SS-16 VOIP-2-GSM package).

Frequency bands	GSM 890-960 MHz PCN 1710-1880 MHz PCS 1850-1990 MHz UMTS/HDSPA 1920-2170 MHz
Gain	2 dBi
Polarization	Vertical
Height	Total 73 mm Active stick 43 mm
Base	Magnetic, diameter 2,8cm
Cable	Type RG174, length 2,5 m
End Connector	Nipple, male

Note:

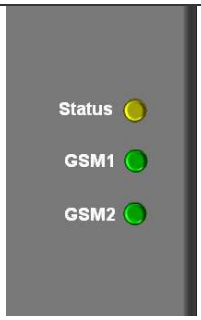
- The SS-16 VOIP-2-GSM unit and its antennas should be placed as far as possible from appliances or office equipment that is sensitive to radio interference (microwave ovens, copiers, TV sets, PC displays, and multimedia systems). For best results, try to find a place of maximum signal reception for the Mobile antenna
- In addition, the antenna must NOT be located near heavy-duty equipment that may generate electromagnetic interferences, such as electric motors or heaters.

5.7 POWER UP

Switch the SS-16 VOIP-2-GSM on - the indicator LEDs will light up.
Remember that the adapter is the disconnection device (there is no POWER switch on SS-16 VOIP-2-GSM), so the 230 V_{AC} socket-outlet must be installed near the equipment and must be easily accessible.

5.8 STATUS INDICATORS

SS-16 VOIP-2-GSM has three optical indicators (bicolour LEDs - red and green or red and yellow) that show the status of the device, respectively of the Ethernet interface and that of the two mobile modules.

The indicators may light up continuously or flash to show data traffic. The three LEDs are located on the upper right corner of the SS-16 VOIP-2-GSM case, above the two slots for SIM cards. They are labelled STATUS, GSM 1 and GSM 2.	 **Figure 6: LEDs location**

Significance of LED indicators:

STATUS: Shows the state of the equipment. Indicates the proper power supply and the state of the Ethernet interface of SS-16 VOIP-2-GSM.

Off = device not powered (no supply voltage), On, Red color = no ETH network is connected, On, Orange color = ETH port connected to the local network, but no traffic, Orange blinking = ETH port connected and data traffic present.

GSM 1: Shows the state of the first mobile module.
Off = not installed (the respective module is disabled from software), red blinking = not registered or SIM not present, lights up green = mobile module1 free (available), lights up red = mobile module1 busy (in a call).

GSM 2: Shows the state of the second mobile module, same significance as first mobile.
Off = not installed (the module 2 is disabled from software), red blinking = not registered or SIM not present, lights up green = mobile module2 free (available), lights up red = mobile module 2 busy (in a call).

6. CONFIGURATION

The SS-16 VOIP-2-GSM can properly perform its functions of high-speed VoIP-2-mobile gateway with the default settings. However, it can be easily configured to meet various usage scenarios. “Configuring” or programming the equipment means adequate setting of all the parameters.
The embedded Linux firmware allows configuring the SS-16 VOIP-2-GSM without the need of additional software on the computer used for programming. A web browser as configuration terminal is all that is needed. This means that the SS-16 VOIP-2-GSM may be used on **any computer platform** and is not restricted to a certain operating system! *The examples in this manual are when using Windows XP SP3, but SS-16 VOIP-2-GSM may be used with any operating system.*
Using the web browser, the configuration can be performed remotely: the desktop PC or notebook may be connected to the SS TELECOMS IP gateway either directly or through a hub by means of wired (Ethernet) connection.

Important Note: *Prior to using this VoIP to Mobile device you should check the basic settings to guarantee it will work in your environment (for instance, it may be required to change the default IP address and subnet mask).*

6.1 REQUIRED CONDITIONS FOR CONFIGURATION

To be able to configure the SS-16 VOIP-2-GSM device, one must be connected to a LAN (Ethernet 10Mb/s or 100 Mb/s) and have access to a computer that is connected

to the same network, via router or switch. Alternately, one may connect the SS-16 VOIP-2-GSM **directl**y to the PC or notebook that is used for programming, either with a crossover cable or with a straight Ethernet cable. The Web interface may be accessed with any Web browser, **no additional software is needed**.

6.2 USING THE WEB INTERFACE

The default address of the webpage of the SS TELECOMS SS-16 VOIP-2-GSM device is https://**192.168.0.225**

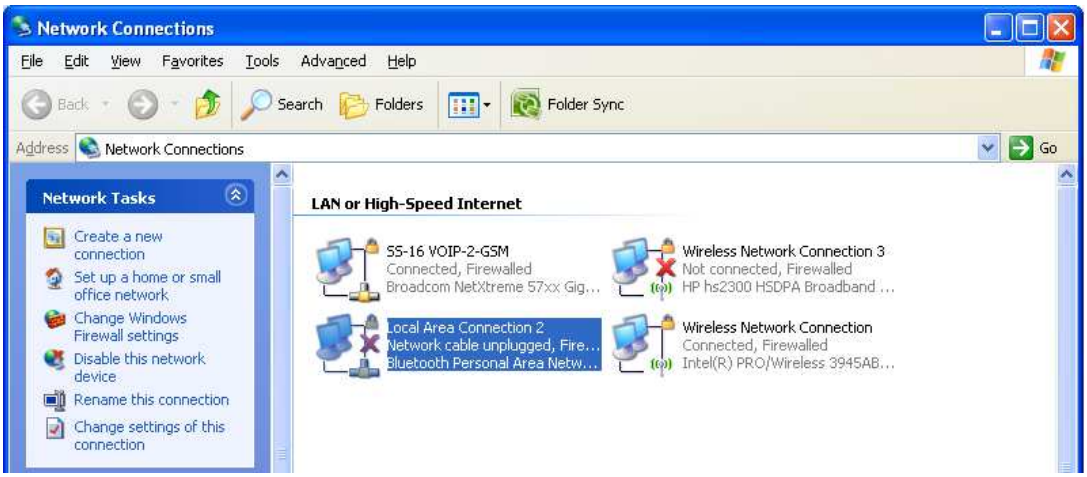
It is recommended to use the default IP address - if the configuration of the local network allows this.

Set up a connection

Establish an Ethernet connection to be used with the SS-16 VOIP-2-GSM unit that needs to be configured.

Go to “Network Connections” on the computer and define a connection to be used for the SS-16 VOIP-2-GSM router.

Use a significant name, such as “SS-16 VOIP-2-GSM” or ”SS Telecoms Gateway” or “VoIP2Mobile”

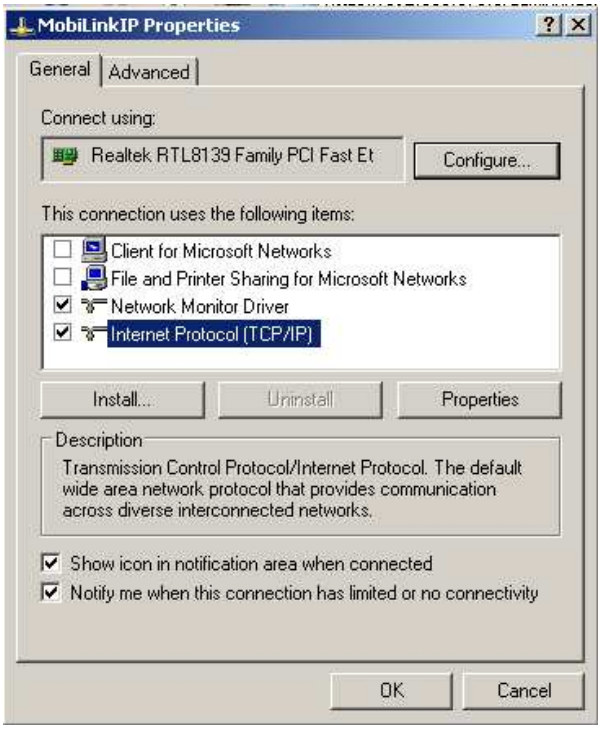


From the network adapters (network cards), select the one which is connected via Ethernet cable to the SS-16 VOIP-2-GSM equipment.

In this example, it is the network board type “Realtek RTL8139 Family PCI Fast Ethernet NIC”

Check the box “Internet Protocol (TCP/IP) and click the button “Properties” to configure your PC.

Configure the connection to SS-16 VOIP-2-GSM

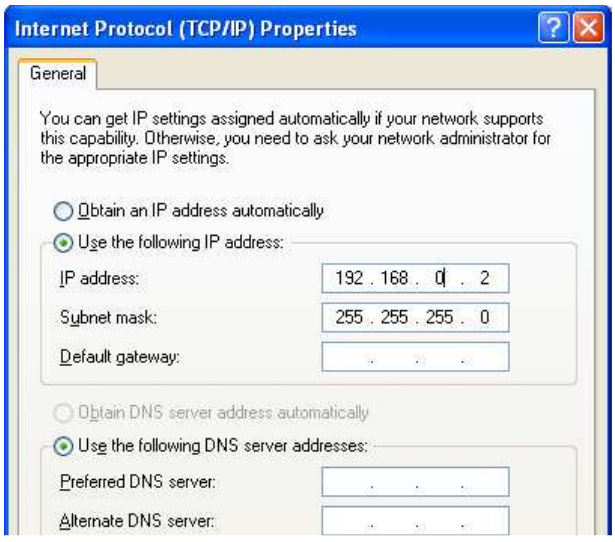


Manual settings

Set the network parameters manually. By default, SS-16 VOIP-2-GSM has the IP address *192.168.0.225* and the Netmask *255.255.255.0*

In case of manual settings, in “Internet Protocol Properties” fill in the corresponding values:

- The IP address could be from 192.168.0.2 up to 192.168.0.254
- The Subnet mask must be the standard: 255.255.255.0
- The Default gateway and “Preferred DNS server” fields may be lefty empty!

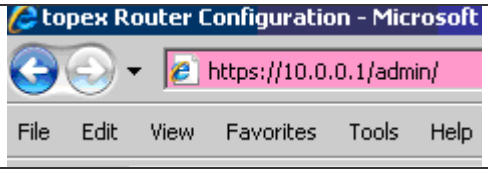

Internet Protocol (TCP/IP) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address: 192 . 168 . 0 . 2
Subnet mask: 255 . 255 . 255 . 0
Default gateway:
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server:
Alternate DNS server:

In the status bar at the bottom of the screen the link icon should be blinking, and a message will show up: “SS-16 VOIP-2-GSM is now connected”

To configure the SS-16 VOIP-2-GSM product using the web interface, just open a favourite web browser and type the default IP address as the URL: **https://192.168.0.225**

These include the IP address 192.168.0.225, allowing you to connect to the SS Telecoms VoIP2Mobile in order to configure it. After reaching the configuration web pages, one may change the IP address of the device according to personal requirements.

For instance, you can use addresses in the range https://172.27.168.xxx/ or **https://10.0.0.1**, as shown in this example:



Hardware Reset – (Factory default settings)

To perform a reset, follow the following procedure because by simply pressing the recessed RST button, a reset will not be performed. Follow these steps:

- shut down the SS-16 VOIP-2-GSM device (unplug the power jack)
- press the RST button and keep it pressed
- power up SS-16 VOIP-2-GSM
- wait until all three LEDs light up in red, and then leave the RST button.

Now the equipment performs a reset and returns to factory default settings. This does NOT happen if the procedure described above is not followed exactly – for instance, if continuing to press RST after the three indicators have turned red, a warning will be given.

Note that the SS-16 VOIP-2-GSM uses a **secure web connection** (https) so one may get several warning messages for example:

Security Alert
Information you exchange with this site cannot be viewed or changed by others. However, there is a problem with the site's security certificate.
The security certificate was issued by a company you have not chosen to trust. View the certificate to determine whether you want to trust the certifying authority.
The security certificate has expired or is not yet valid.
The name on the security certificate is invalid or does not match the name of the site
Do you want to proceed?
Yes
No
View Certificate

or similar warnings mentioning “Website Certified by an Unknown Authority”:

Website Certified by an Unknown Authority
Unable to verify the identity of Topex as a trusted site.
Possible reasons for this error:
- Your browser does not recognize the Certificate Authority that issued the site's certificate.
- The site's certificate is incomplete due to a server misconfiguration.
- You are connected to a site pretending to be Topex, possibly to obtain your confidential information.
Please notify the site's webmaster about this problem.
Before accepting this certificate, you should examine this site's certificate carefully. Are you willing to to accept this certificate for the purpose of identifying the Web site Topex?
Examine Certificate...
Accept this certificate permanently
Accept this certificate temporarily for this session
Do not accept this certificate and do not connect to this Web site
OK
Cancel
Help

Click „Yes", „OK" or „Accept" to continue.

Type "OK" to accept the certificate for the SS-16 VOIP-2-GSM website. Other "Security Error" messages may warn about "Domain Name Mismatch", referring to the security certificate. Again click "OK" to continue.

File Edit View Favorites Tools Help
Certificate Error: Navigation Blocked
There is a problem with this website's security certificate.
The security certificate presented by this website was issued for a different website's address.
Security certificate problems may indicate an attempt to fool you or intercept any data you send to the server.
We recommend that you close this webpage and do not continue to this website.
Click here to close this webpage.
Continue to this website (not recommended).
More information

One may receive the warning "You have requested an encrypted page" or "You are about to view pages over a secure connection".

This is normal; it shows connection to SS-16 VOIP-2-GSM over a secure, encrypted link.

Security Alert
You are about to view pages over a secure connection.
Any information you exchange with this site cannot be viewed by anyone else on the Web.
In the future, do not show this warning
OK
More Info

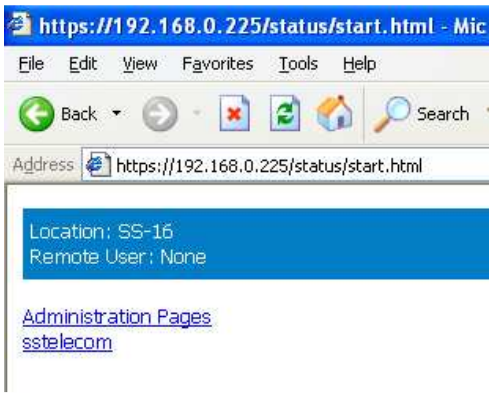
The secure connection is confirmed by the “padlock” symbol that shows up in the status bar at the bottom of the screen, indicating a secure (encrypted) connection. SS-16 VOIP-2-GSM uses 128-bit SSL encryption to prevent hackers from capturing passwords and sensitive data which is the same security used by banks and the military.

or

Internet Explorer shows the padlock icon only when everything on the entire webpage is encrypted. But even when the browser doesn't show a padlock, the "https://" address means the data is still encrypted.

Other browsers, such as Mozilla Firefox, will also show a lock icon near the SS-16 VOIP-2-GSM connection, together with a message that says: “Authenticated by SS Telecoms”.

- Administration (configuration pages)
- SS Telecoms webpage



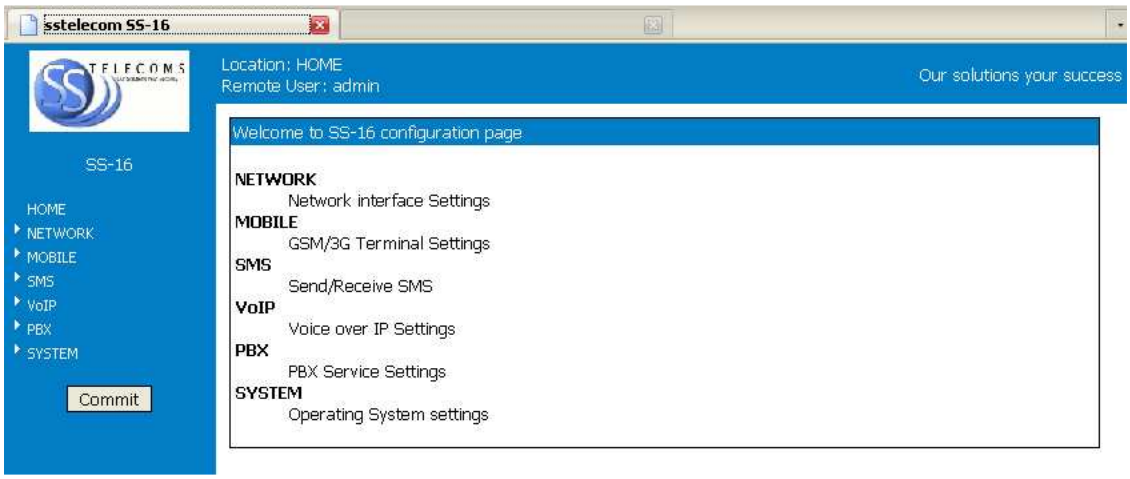
Click the link of interest.

Now the user will be asked to enter a user name and a password to access the configuration page of SS Telecoms SS-16 VOIP-2-GSM.

For the administrator of the system network, the default user name is **admin** and the password is **99admin11**. Later, this password may be changed using the web configuration page, as described in the paragraph about Password.

For security reasons, it is strongly recommended to change the default password with one of own choice as soon as possible Also, do NOT mark the checkbox “Remember my password” in the browser

Following successful log-in, the main administration page for the SS TELECOMS SS-16 VOIP-2-GSM VoIP gateway should be shown on screen.



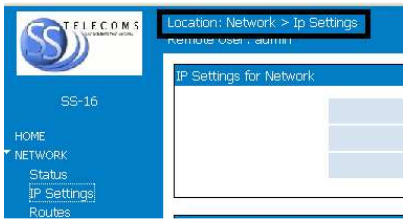
Menu items

There are several sections (Menu items) on the configuration page of SS-16 VOIP-2-GSM, allowing modifying the settings for:

- **HOME**: this home page for configuration of the equipment.
- **NETWORK**: settings for the local wired (Ethernet) network
- **MOBILE:** parameters for the 2G or 3G mobile voice / data networks
- **SMS:** To configure or send and Receive SMS's / E-mail to SMS / SMS to E-mail
- **VoIP:** parameters related to the Voice over IP functions;
- **PBX:** Settings for the private phone exchange features;
- **SYSTEM:** Performs operating system functions (update, change of password, logs) and Shows current status

Depending upon the actual firmware version running on the SS-16 VOIP-2-GSM gateway, one may or may not have access to all these configuration sections or sub-sections (features, services).

The Menu bar is located to the left, and features the button Commit at the bottom.
On top of the web page, next to the SS Telecoms logo, is a "Location" indication, which reminds the user exactly where he is on the Web interface. In this example, the section is **Network** and the subsection **IP settings** (IP Settings for the Ethernet connection).



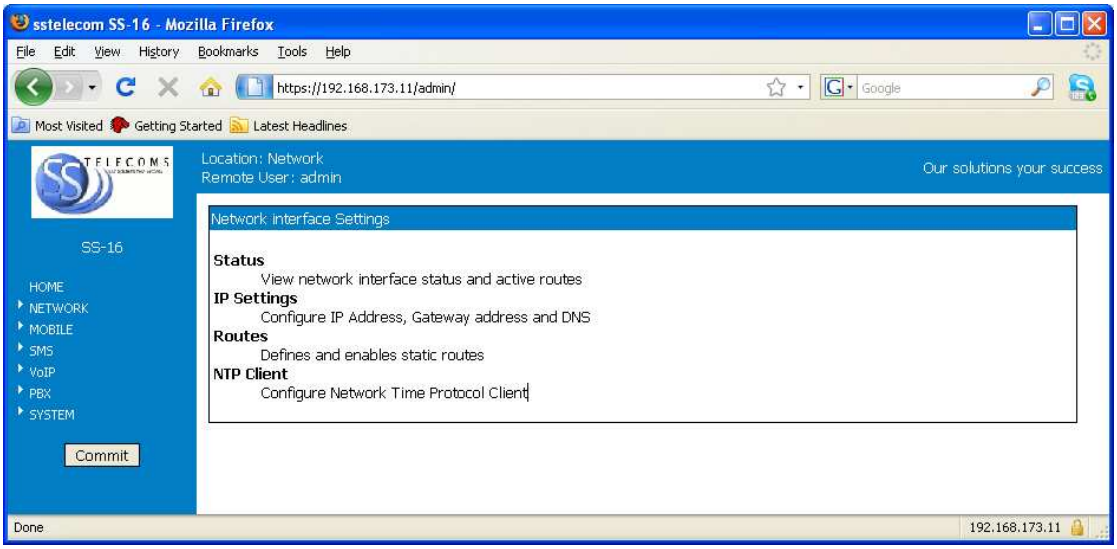
Note:
- There are two types of pages, 'Status' and 'Settings'
- At the bottom of every 'Settings' page are these three buttons:

The button **Save** is used to save the changes performed, the **Cancel** button aborts the changes (closes the window without modifying the current settings), and the **Default** button will reset the values in the current page to the default (initial) configuration.

When changing the settings in any page, press the "Save" button keep these settings. Otherwise, clicking any other link, reaching another configuration page, the modifications performed will be lost.

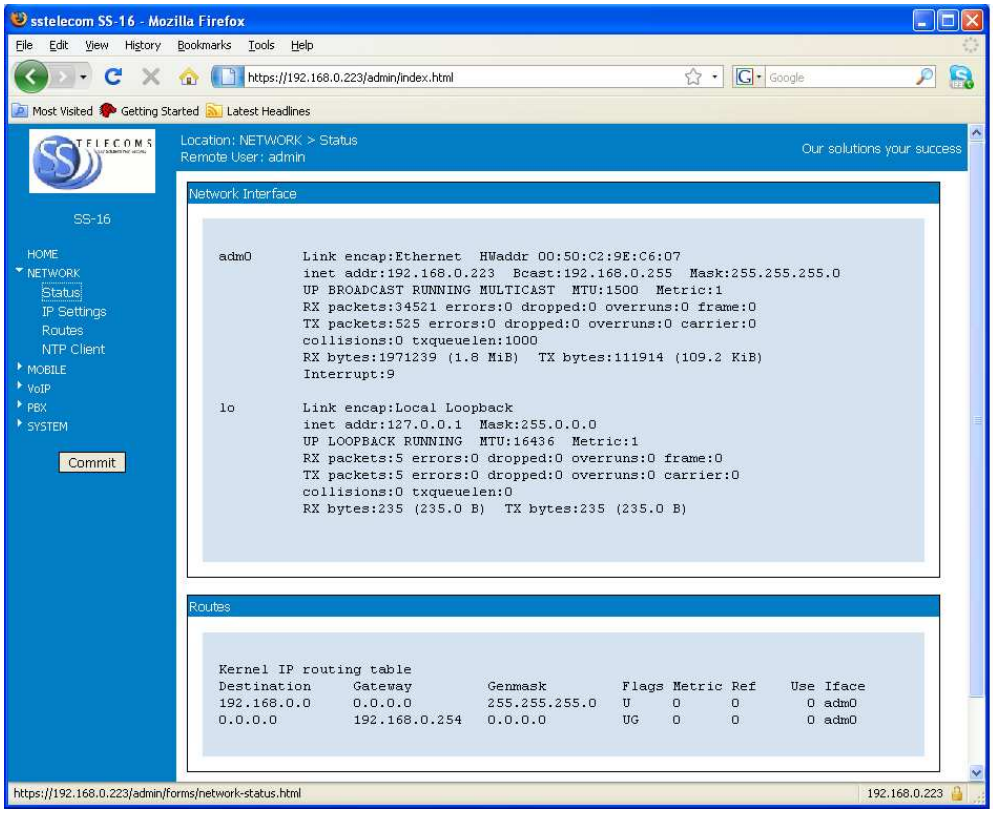
6.3 NETWORK

This group features the pages of settings related to the local wired Ethernet network, for IP, routes, and respectively for the NTP Client:



6.3.1 Status

Each configuration submenu begins with a "Status" page that shows the current state for the respective group of parameters.
It this case, it shows info about the state of the Ethernet interface and the current static routes:



Network Interface
This is the detailed information about the Ethernet network interface of the SS-16 VOIP-2-GSM. It contains two similar groups of network parameters:

Adm0

```
adm0      Link encap:Ethernet  HWaddr 00:50:C2:92:CC:4A
          inet addr:192.168.173.4  Bcast:192.168.173.255  Mask:255.255.0.0
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:1163193 errors:0 dropped:0 overruns:0 frame:0
          TX packets:34338 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:66834569 (63.7 MiB)  TX bytes:3935432 (3.7 MiB)
          Interrupt:9
```

Details about the real Ethernet interface, including MAC (physical address), Internet address (logical address), Broadcast, subnet mask, MTU value, metric, statistics (packets, errors, collisions, mega bytes transferred) for the reception (RX) and respectively transmission (RX).

lo

Local Loopback interface

```
lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:886 errors:0 dropped:0 overruns:0 frame:0
          TX packets:886 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:43809 (42.7 KiB)  TX bytes:43809 (42.7 KiB)
```

The same parameters are shown for the virtual, local loopback interface, at the standard address for loopback, “127.0.0.1”. Since it is a loopback interface, the number of bytes sent out will always be identical to the number of bytes received.

Routes

Shows the current routes, taking the information from the Kernel routing table for IP

Routes

```
Kernel IP routing table
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
172.168.0.0     192.168.1.1     255.255.0.0     UG    0      0        0 adm0
192.168.0.0     0.0.0.0         255.255.0.0     U     0      0        0 adm0
0.0.0.0         192.168.1.2     0.0.0.0         UG    0      0        0 adm0
```

Or:

Routes

```
Kernel IP routing table
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
192.168.0.0     192.168.1.2     255.255.0.0     UG    0      0        0 adm0
192.168.0.0     0.0.0.0         255.255.0.0     U     0      0        0 adm0
0.0.0.0         192.168.1.2     0.0.0.0         UG    0      0        0 adm0
```

The Static Route Display shows the current routing table of the SS-16 VOIP-2-GSM

Note that this is the Status page for network - the existing static routes for the SS-16 VOIP-2-GSM (default routes, gateway, masquerading if used, flags, metric, interface used, etc) can be seen and changes cannot be performed.

If needing to change the routes, go to the section 'Routes' of this page, for defining and modifying routes.

The routes are shown from the top downwards from the most specific to the least specific.

Routes

```
Kernel IP routing table
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
89.234.0.0      192.168.144.253 255.255.255.0   UG    0      0        0 adm0
10.0.0.0        192.168.1.8     255.255.0.0     UG    0      0        0 adm0
192.168.0.0     0.0.0.0         255.255.0.0     U     0      0        0 adm0
0.0.0.0         192.168.1.2     0.0.0.0         UG    0      0        0 adm0
```

Destination - the destination network or host system. Declaring a default route means adding a route with 0.0.0.0 as its destination. In case of networks, the last two groups of figures contain only the digit zero.

Gateway - the access gateway the respective routing entry points to. If this field has only zeroes (0.0.0.0) or an asterisk (*), this means no gateway is used, the destination network is connected directly to the computer.

Genmask - the bit mask applied to the destination. It shows the "generality" of the route.

Flags - indicator flags that describe some characteristics of the route. Such flags are:

U - the route is active and operating, the interface to be used is up;

G - the route uses an external gateway (the network interfaces of the system supply routes to the networks which are directly connected, all other routes use external gateways. Thus, the networks which are directly connected don't have the G flag, but it is activated for all other routes).

H – it is a route towards a single host instead of a network.

D - the table entry has been generated dynamically, either by a routing protocol or by an ICMP redirect message (see section 3.5).

M – this route was modified by a dynamic routing protocol;

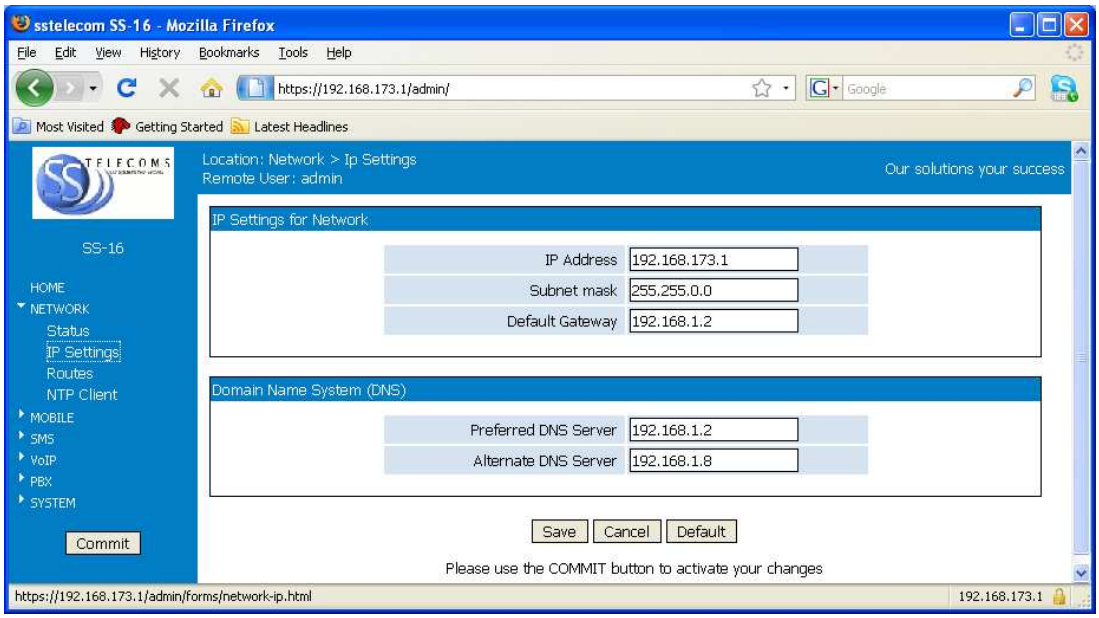
R - the route was re-activated following update by dynamic routing protocol. Routes may be configured as passive or static even when a protocol for dynamic routing is used.

Other fields refer to the **Metric** (routing cost), the number of references to this route, etc,

Finally, **Iface** is the name of the network interface used for this route.
For the Ethernet interfaces there is currently a single name, respectively adm0.

6.3.2 IP Settings

Here one can set up the parameters for the equipment in the local network.
Settings for the LAN IP of the SS-16 VOIP-2-GSM gateway are the standard ones: IP Address and Netmask, Gateway (if used), Preferred and Alternate server for DNS.



These settings control how the SS Telecoms equipment connects into the local wired (Ethernet) computer network.

IP settings for network

The default address value is 192.168.0.225 and the netmask is 255.255.255.0.

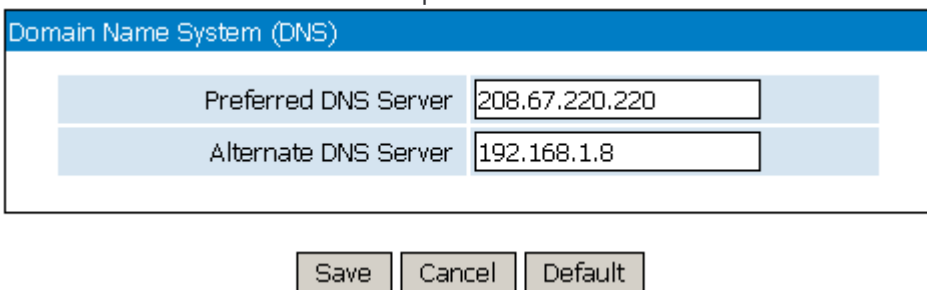
One may change these default settings, if the configuration of the network requires this, for instance to *10.0.0.1* in this example:	IP Address 10.0.0.1 Netmask 255.255.255.0
Usually the IP address allocated for SS-16 VOIP-2-GSM on the LAN side is a **non-routable** (internal) address, as the ones shown here, in the range 192.168.xxx.yyy:	Location: Network > Ip Settings Remote User: user Empowering Communications IP Settings for Network IP Address 192.168.173.4 Subnet mask 255.255.0.0 Default Gateway 192.168.1.2

When changing this IP address, SS-16 VOIP-2-GSM tries to redirect the web browser to the new address automatically. If it does not succeed, update the configuration page by entering the new IP address in the browser, such as https://192.168.173.4 in the above example and pressing the Reload button of the browser.

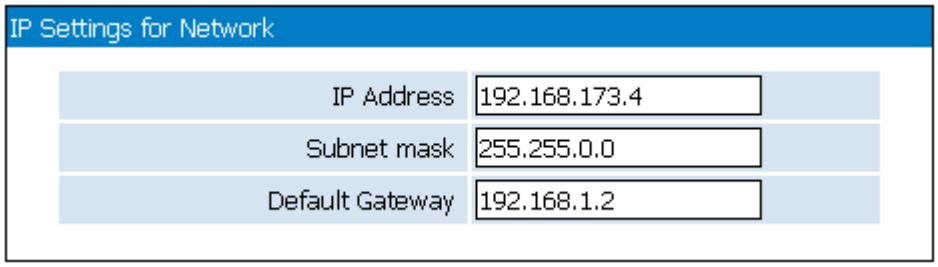
DNS Servers:

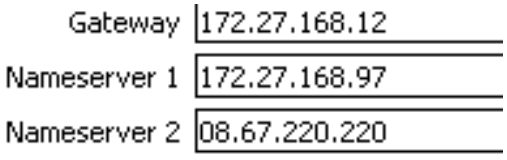
Select which primary and secondary servers for DNS (Domain Name Service) will be used.

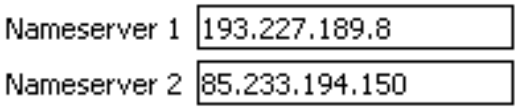
The Name Servers may be from the local network (address 192.168.x.y) as the second entry in the following example or from the public network, the great Internet, such as 208.67.220.220 in the example below:



Other name-server examples







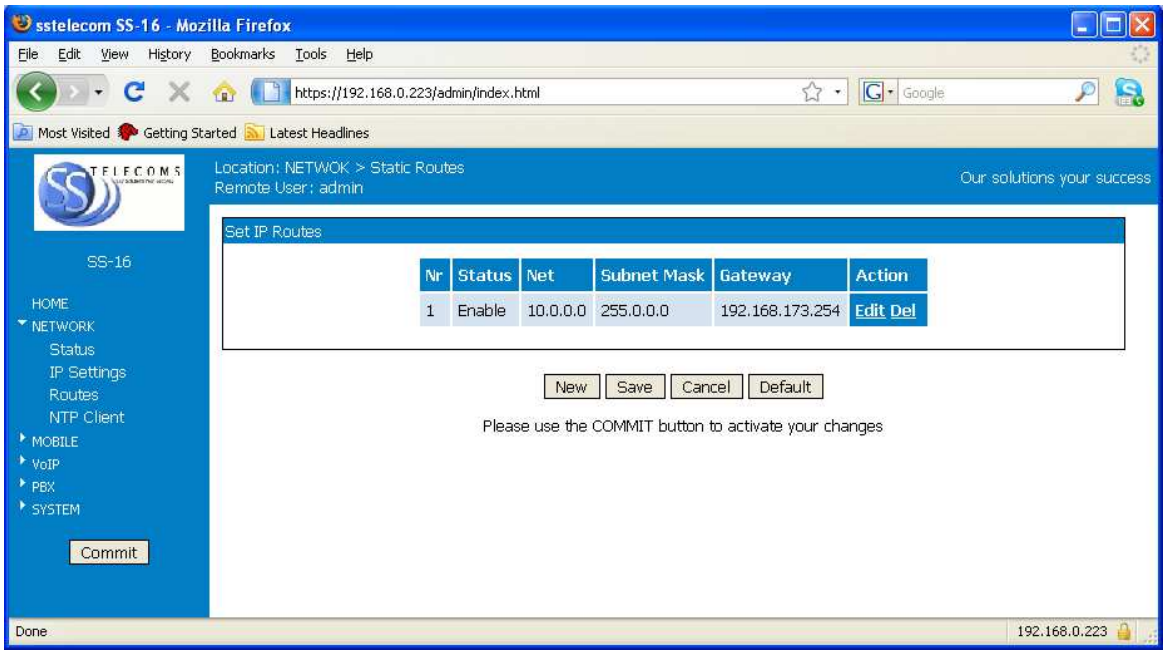
Saving

Save Cancel Default	Always use the “Save” button to save the settings, then “Commit” (down on the column to the left of the screen) to make permanent the IP settings.

6.3.3 Static Routes

Routing means determining and prescribing the path or method used for forwarding data packets. This option page is concerned with defining fixed (static) routes. It shows the current routing table for SS-16 VOIP-2-GSM and allows defining several static routes. The routes established here will be shown in the page Network ->Status-> Routes.

"Set IP Routes" lets one edit, remove or add routes in the table shown below:

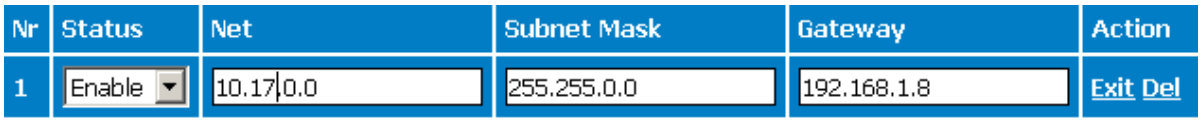


Static Route Definition

Such a static route is a possible path from a device to its destination or to another host. One **must** insert predefined rules of routing for the SS-16 VOIP-2-GSM in case of appending one or several network devices such as routers behind the SS-16 VOIP-2-GSM device, to share the same connection to the mobile Internet. This way the SS-16 VOIP-2-GSM router will be able to know where it may deliver the data packets coming from the Internet with different destination IP addresses.

Buttons

Use **New** to create a new row (route definition), **Edit** to edit an existing definition, **Exit** get out of the table, and **Save** to save the changes. **Del** erases from the table an existing route.



Defining a route

Net: Address of the remote network or host to which the user wants to assign a static route. Notice that it is not an IP address, like the one for the SS Telecoms SS-16 device, but the address of a network, so that the last two groups of figures will always be zero.

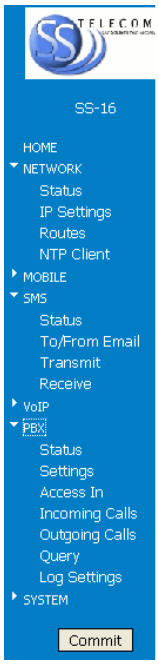
Netmask: the subnet mask determines which portion of the destination IP address is the network part and which is the host part.

Gateway: the gateway to be used, enter here the IP address of the router which allows for contact between SS-16 VOIP-2-GSM and the remote host or network.

Enable:

Each of the routes defined or edited can be individually enabled or disabled.
This is especially useful when configuring several SS-16 VOIP-2-GSM devices one after another, or when the configuration and requirements of the local network change frequently.
In this case pre-define several routes, even if they are not currently in use, and leave enabled only the ones strictly necessary for the current configuration.
The other routes are available; they may be edited and enabled as the need arises.

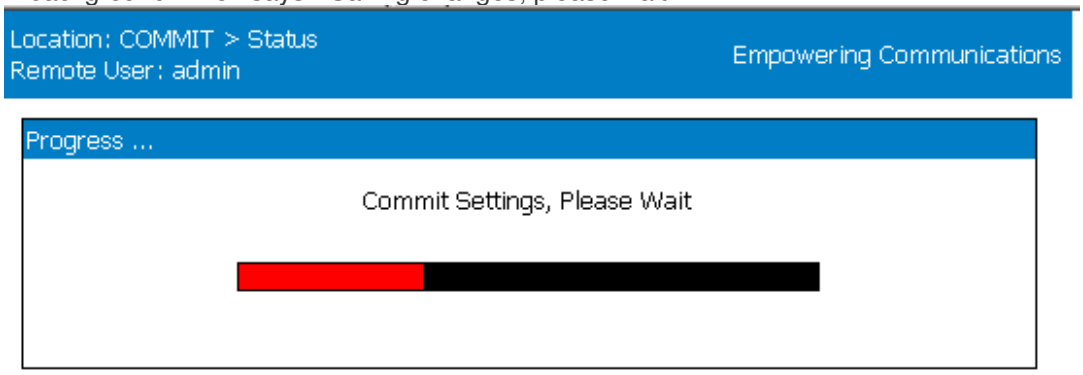
6.3.4 Commit



The “Commit” option described here is not specific to the Network, but it is rather a general option for SS Telecoms devices.

If configuration changes were performed, using the “Save” button is not enough since it saves the modifications only into the temporary memory of the SS-16 VOIP-2-GSM.

Always press the “Commit” button if these changes need to be permanent.
This button is located at the bottom of the Menu list, to the left side of the screen.
There is a “Commit Settings” message and, underneath it, a red progress bar over black background which says: “Saving changes, please wait …”

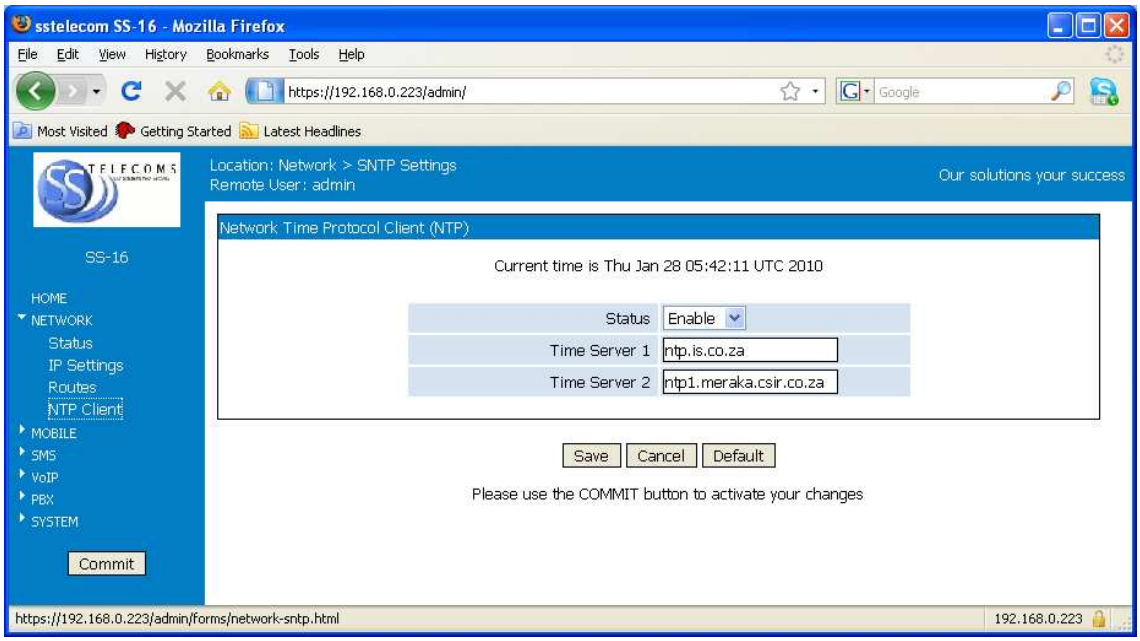


Saving will take some time, because the changed settings are saved into permanent (Flash) memory of the equipment.
The SS-16 VOIP-2-GSM resets, and when it starts again, the new set of parameters will be active.

Warning: While committing changes, when resetting the SS-16 VOIP-2-GSM or while loading a new program image, the equipment will cease operation for a few seconds. This means all connections: voice, local network etc. will be interrupted, but will resume when the SS-16 VOIP-2-GSM starts again.

6.3.5 SNTP Client

Enables and configures the NTP Client of SS-16 VOIP-2-GSM.



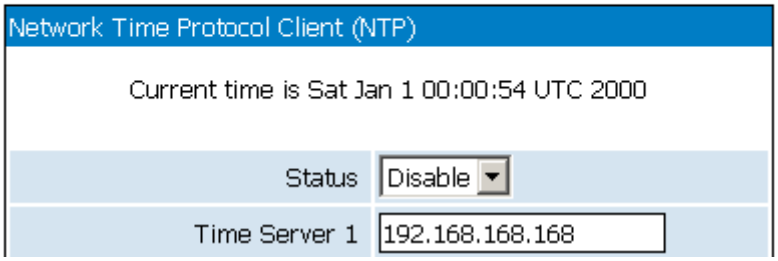
The **Simple N**etwork **T**ime **P**rotocol is used to update the real-time clock in a computer, over Internet.
It uses one or several remote dedicated Time servers on the Internet that accurately synchronizes the system date and time.

Status of NTP Service: Enabled or Disabled; by default it is disabled

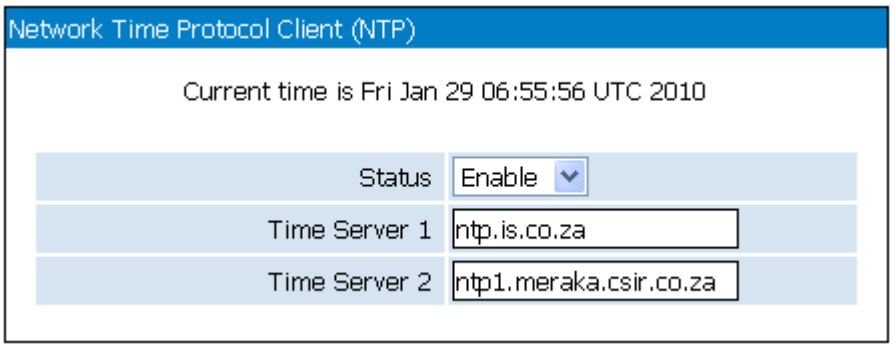
Server: enter the full name of the NTP server in need to use, such as “time.nist.gov” or “ntp.is.co.za”.

The “Current time” indicator shows the current UTC time (Universal Coordinated Time) of the system, with millisecond precision.

If NTP feature is not active (it is disabled, the selected server is not online, or there is no connection to the Internet) it will display the default UTC date/time, which is begins at Jan 1, 2000 and 0:00:00.



When the NTP Client becomes active, it shows the current UTC time:



This UTC time may be different from the local time of your computer!

In the system logs one can see the moment when the applications running on SS-16 VOIP-2-GSM connects to the Internet and the time jumps from the default “Jan 1 2000” to the actual date/time supplied by the NTP server selected by the user:

```
Jan  1 00:00:08 (none) kern.warn kernel: udp redirect init
Jan  1 00:00:08 (none) kern.warn kernel: udp_redirect proc initialization okay!
Jan  1 00:00:08 (none) kern.warn mini_httpd[101]: started as root without requesting
chroot(), warning only
Jan  1 00:00:08 (none) kern.notice mini_httpd[101]: httpd starting on bytton, port 443
Jan  1 00:00:08 (none) kern.warn kernel: IFX TAPI v3 - The Common Voice API, version
3.5.1.4
Jan  1 00:00:08 (none) kern.warn kernel: ^M(c) Copyright 2006, Infineon Technologies AG
Jan  1 00:00:08 (none) kern.warn kernel: IFX VINETIC device driver, version 1.2.6.4
Jan  1 00:00:08 (none) kern.warn kernel: ^M(c) Copyright 2006, Infineon Technologies AG
Jul 10 13:18:14 (none) user.notice root: Receive Time form 192.168.168.168
Jul 10 13:18:15 (none) kern.warn kernel: reg_buffer_pool_get is called
Jul 10 13:18:15 (none) kern.info kernel: reg_callback_bufferpool_get, handle 811FA840
Jul 10 13:18:15 (none) kern.debug kernel: QOS init at highlevel.
Jul 10 13:25:52 (none) user.info /mnt/app/bin/cenq: Command: view portsoncard 0
```

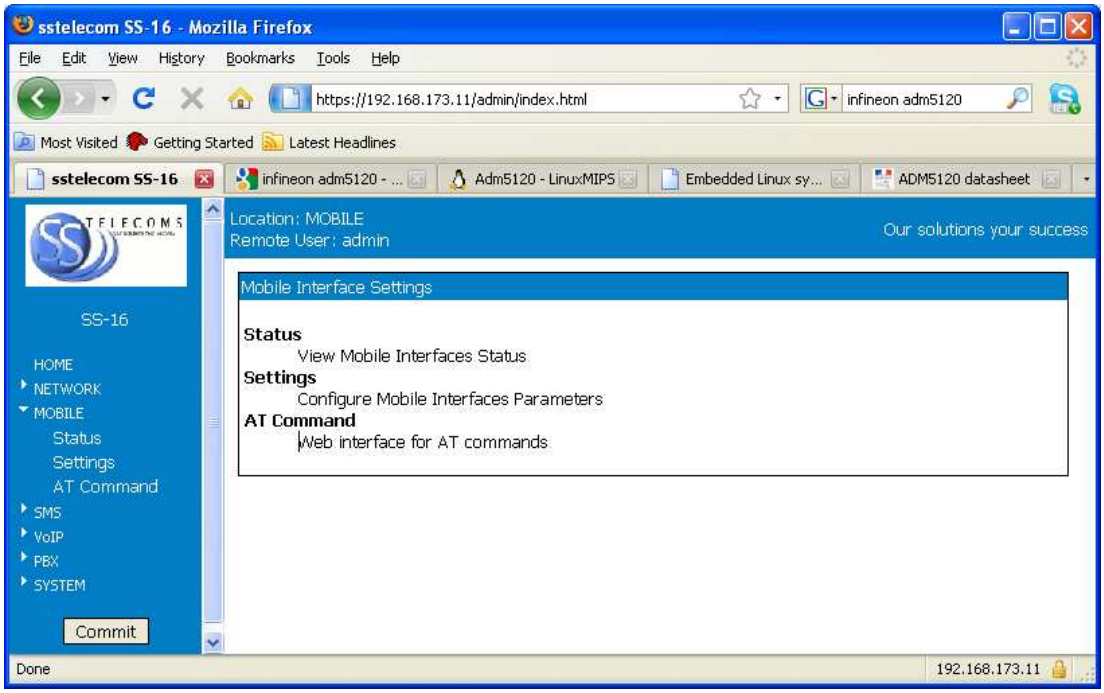
Or:

```
Jan  1 00:00:08 (none) kern.warn kernel: udp redirect init
Jan  1 00:00:08 (none) kern.warn kernel: udp_redirect proc initialization okay!
Jan  1 00:00:08 (none) kern.warn kernel: IFX TAPI v3 - The Common Voice API, version 3.5.1.4
Jan  1 00:00:08 (none) kern.warn kernel: ^M(c) Copyright 2006, Infineon Technologies AG
Jan  1 00:00:09 (none) kern.warn kernel: IFX VINETIC device driver, version 1.2.6.4
Jan  1 00:00:09 (none) kern.warn kernel: ^M(c) Copyright 2006, Infineon Technologies AG
Jul 20 13:18:20 (none) user.notice root: Receive Time form 192.168.168.168
Jul 20 13:18:20 (none) user.info bytmspd: bytmspd.c: *** bytmspd run! ***
Jul 20 13:18:20 (none) user.info bytmspd: slotvin.c: init slotvin nr=2
Jul 20 13:18:20 (none) user.info bytmspd: chipvin.c: Init chip vinetic &bytvin[0]=0x10015300
Jul 20 13:18:20 (none) user.info bytmspd: chipvin.c: Open Device /dev/vin10
```

Don’t forget to click the Save button and then “Commit” to make the change performed permanent.

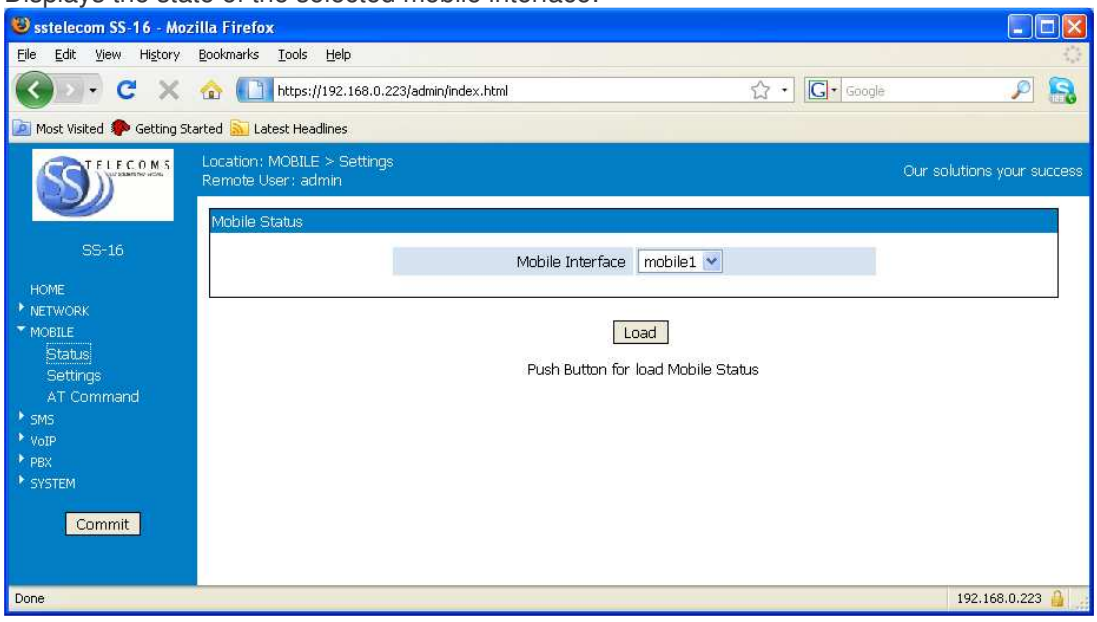
6.4 MOBILE

The “Mobile” section is for viewing and configuring the two mobile interfaces of the equipment.
It also has two sections, one for displaying the current state, and the other for configuring the mobile modules.

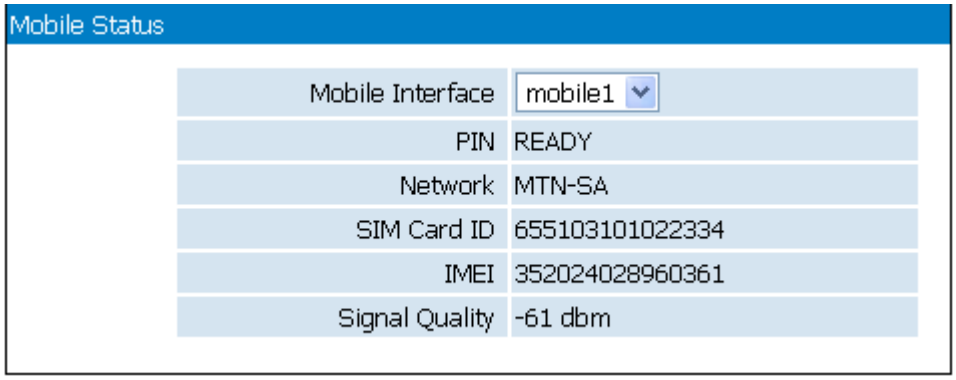


6.4.1 Status

Displays the state of the selected mobile interface:



First, select the Mobile interface whose state will be shown – select 1 or 2 , then press “Load” to see the current state (below for Mobile 1):



Likewise, should “Mobile2” be selected, the SS Telecoms device will show the state of the other GSM module, in this example registered to a different carrier:

PIN – state of the PIN code. **Ready** is the normal state, which means the SIM is active (unlocked) and the equipment may be used. It happens when the PIN is not requested, or that it is asked for, the correct value has been entered, so the Sim can operate. If it shows WRONG, it means the respective SIM is blocked because of a missing or incorrect PIN code. Go to the Mobile settings configuration page and input the correct PIN to unlock the subscriber card.

Network – when the respective Sim is properly registered, this field shows the name of the mobile network where the equipment is registered, as transmitted by the mobile carrier and understood by the mobile. If the Sim is missing or not properly registered, this field may display “Limited service”

Sim card ID – it shows in fact the IMSI (International Mobile Subscriber Identity) code, that is specific to the subscriber, therefore to the Sim card used in the SS Telecoms equipment.

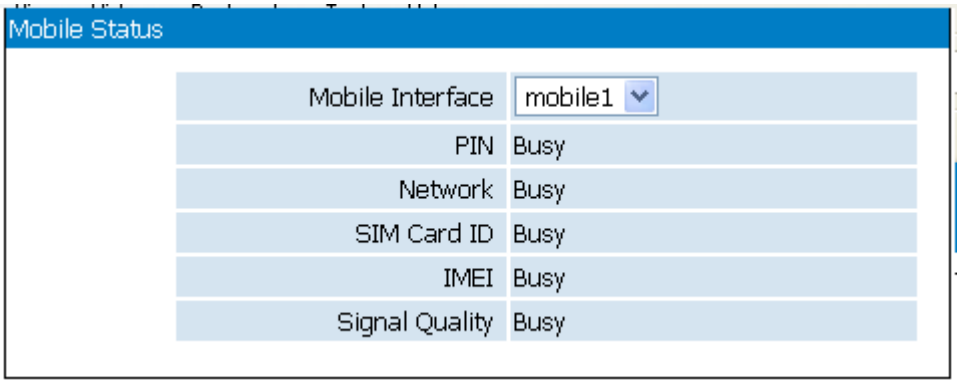
If the Mobile module of SS-16 detects an active SIM card, it will query (and display) the IMSI code. Since the IMSI code is related to the SIM, when changing the SIM card used with the respective GSM module, the user will have another IMSI code.

IMEI - the International Mobile Equipment Identifier is a 15-digit number that uniquely identifies an individual mobile terminal wireless device. While IMSI is specific to the subscriber (SIM card), IMEI is specific to the equipment (cellular modem of SS-16 VOIP-2-GSM)
If the SIM card is missing, the SIM Status window will show “error” in the fields PIN, IMSI, Signal Level, Network and respectively Registration.
But the user will still see the IMEI code, since it is equipment-dependent, and not related to the SIM card.

Signal Quality - The level of the signal received from the mobile network.

At the bottom there is a “Load” button again - click it to refresh the display of module’s status. This way one is sure to see the current state of the mobile module and subscriber information!

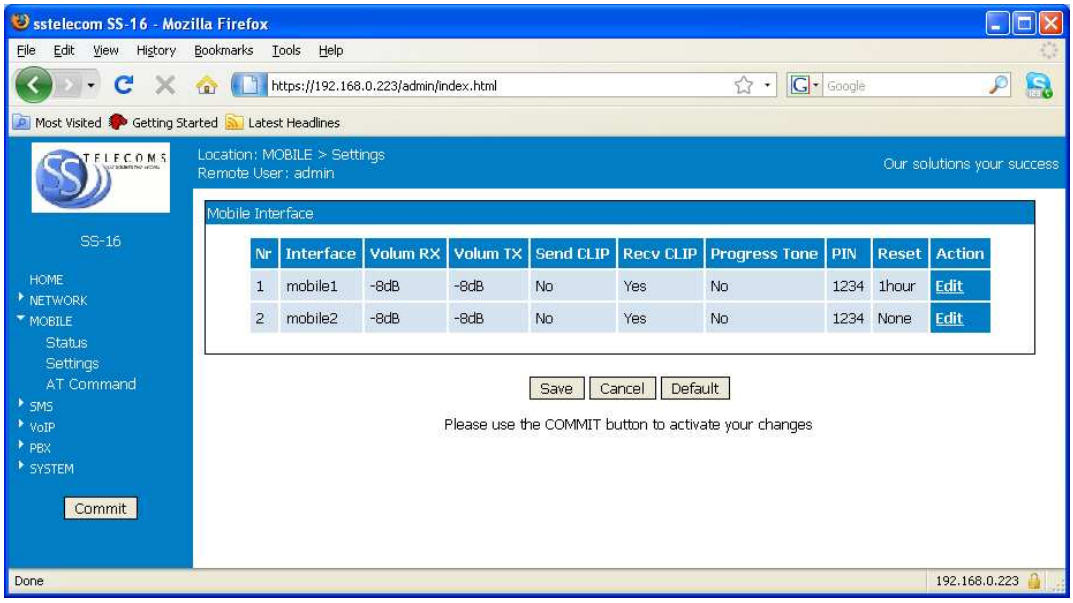
Notes:
To be able to display the current state, the respective mobile module must **NOT** be in a call!
When the GSM module is routing a VoIP call, or transmitting an SMS message, it will no longer be available for status **interrogation**, so it will display “Busy”:



*This does not mean something is wrong, simply that the module is unavailable for status **interrogation**.*
Try again (click "Load") later, when finished with the voice call or the transmission of SMS messages.

6.4.2 Settings

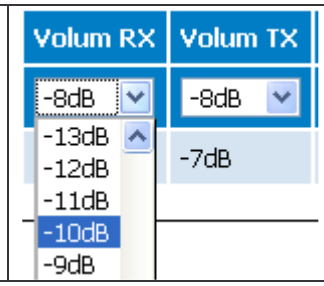
The settings for the two Mobile interfaces: One can edit the parameters for each of the two cellular interfaces of the SS-16 VOIP-2-GSM.



Click "Edit" of the Mobile Interface where you want to change the parameters:

Volume

One can modify the sound level both for the output – TX and input – RX of the voice channel. Select Volume to change the audio level for **input** and **output.** The default is -8 db.



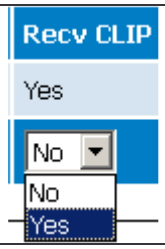
One may use the "Default" button located at the bottom of the page to restore the default level for both input and/or output.

Definition of RX and TX
In this page for Mobile Settings, RX is the voice path coming form the mobile network towards VoIP, while TX is the reciprocal voice path, which comes from VoIP towards the mobile network.
Note1: Don't forget to use the button "**Save**" to store the changes performed over the audio volume.
Note2: The audio volume can be set up only when the State of the Mobile module is ready (registered but not busy).
If the module was busy dialling a number, or sending / receiving SMS messages, it is not available for **interrogation**, so the request to set a different audio level will not be applied.

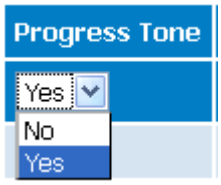
Note3: Do *not change the audio level parameters unless necessary, as this may affect the voice quality.*

Send CLIP – one may select to **send out** or not the Caller ID (presentation). If choosing No, then it will perform **CLIR** - the identity of the call will **not** be sent (identity restricted). The default value is Yes. Several services, including routing of calls, rely on the ID of the caller, if selecting No, these services may no longer be accessed.

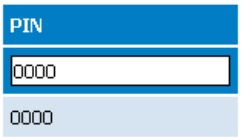
Receive CLIP – the same applies for **receiving** the *calling line identification presentation.*
Note – *the options concerning CLIP enable or disable sending or receiving of ID for **all** the calls performed over the respective Mobile module*



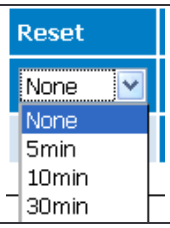
Progress Tone
Enables or disables Call Progress Tones. Call Progress Tones are generated at the beginning of a mobile originated call setup.



PIN
SIM PIN Number required when Pin request is enabled on the SIM.



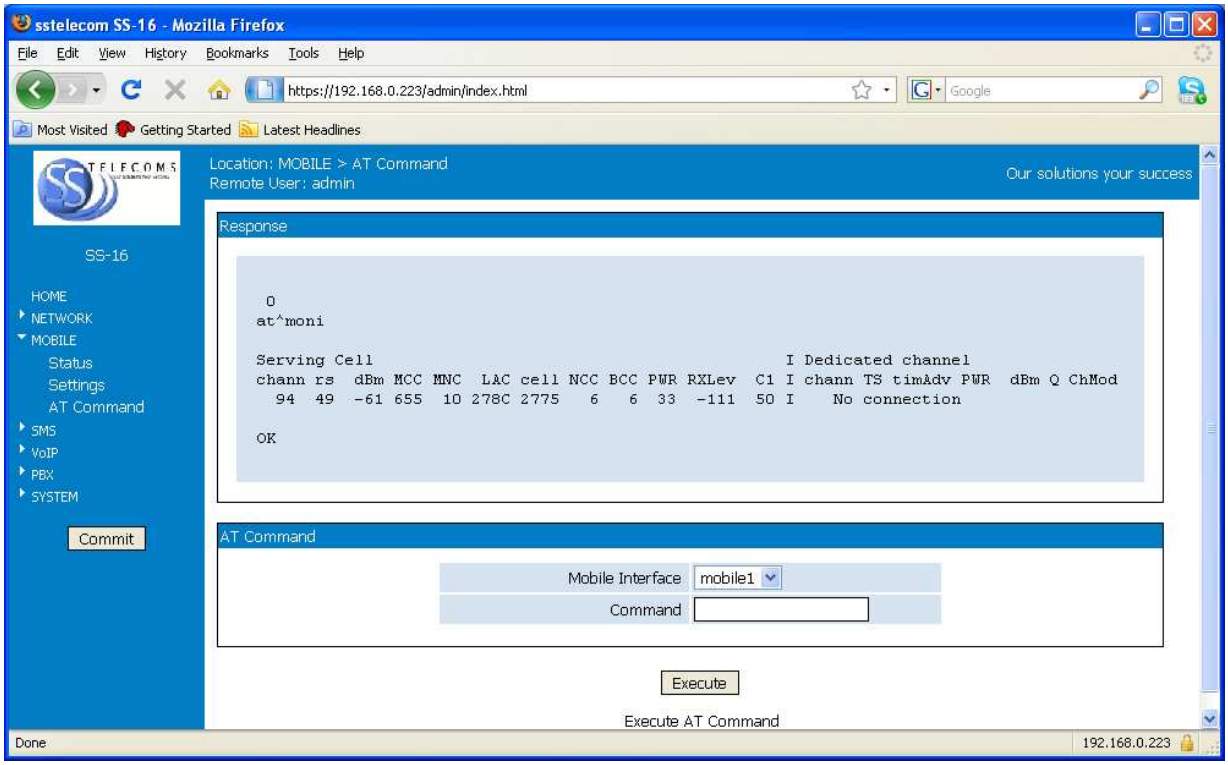
Reset
Set the interval between forced network log-offs.



Action
The last field, Action, selects the action to be performed.
First, "**Edit**", to choose the row (Mobile module) where settings needs to be changed. Now the field name changes to "**Exit**", which should be clicked to leave the respective row of settings. When finished configuring, don not forget to press Save before leaving the page.

6.4.3 AT Command

Allows you to send AT commands directly to the GSM modules and see the respective results:



First select the GSM module to be queried, choose mobile Interface 1 or 2.
Then type in the field “Command” the AT command that you want to issue to the module, such as “AT^MONI” which displays the type of mobile module, in the above example, or “AT+cpin?”, and press the button “Execute” to execute the command.
The command and the result (the response from the module to the query) will be displayed in the upper field, named “Response”, as you can see in the examples below for “AT+cops?”,:

The Response field shows:
- which mobile interface has been queried, by displaying a number above the command and the answer to the command, respectively 0 (for mobile 1) or 1 (for mobile 2)
- the command issued
- the response
- the result, either “OK” or “ERROR”.

AT Commands

AT commands are instructions used to control a modem. Many of them, initially used to control wired dial-up modems, are also supported by 2G/3Gmodems and mobile phones. Besides this common (basic) AT command set, GSM/GPRS modems and mobile terminals support an extended AT command set, which is specific to the GSM technology. The SS-16 internally use extended commands for voice, data and SMS, but with this configuration page, you can send commands directly to the wireless modems of the equipment.

Some examples of such commands are shown below:

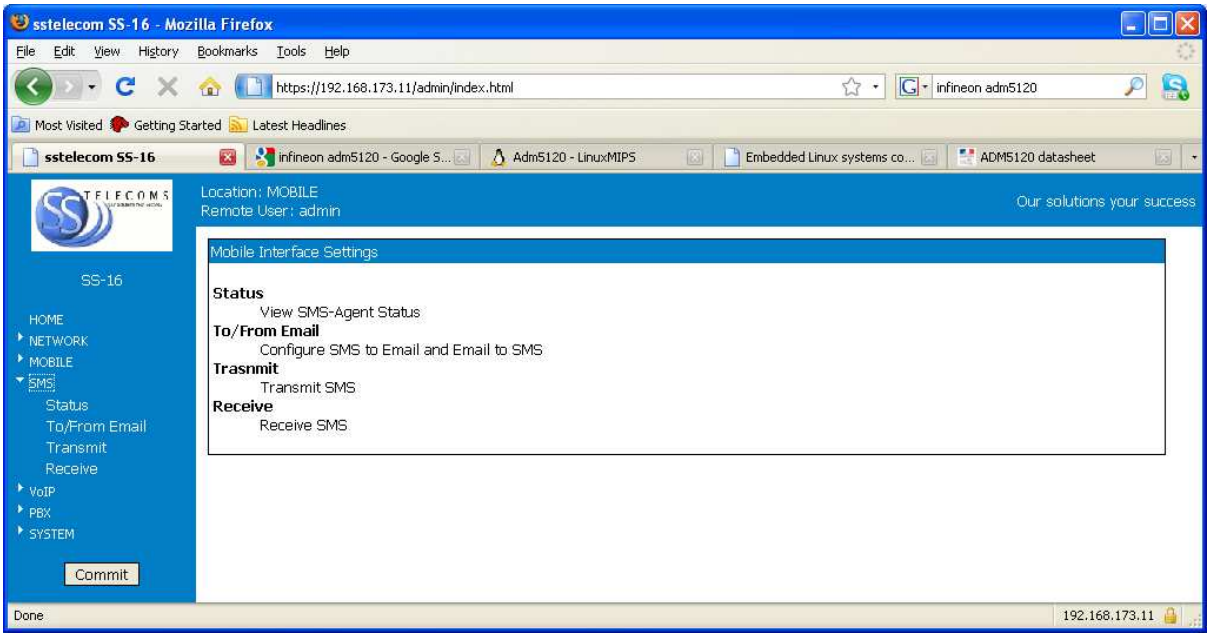
- Get basic information about the mobile modem. Such information may be the name of manufacturer (AT+CGMI), model number (AT+CGMM), IMEI number (International Mobile Equipment Identity) (AT+CGSN) and software version (AT+CGMR).

0 AT+CGMI SIEMENS OK	0 AT+CGMM MC55i OK	0 AT+CGSN 352024024759098 OK	0 AT+CGMR REVISION 01.003 OK

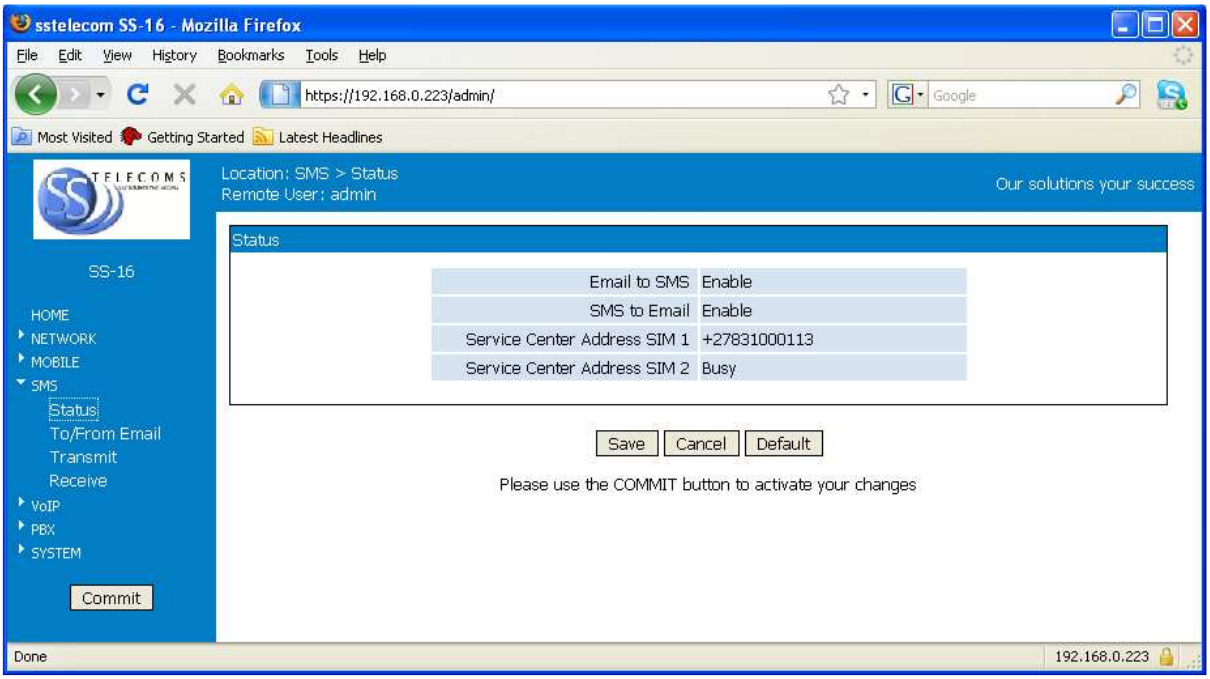
• Get basic information about the subscriber. To find out MSISDN use AT+CNUM and for the IMSI code (International Mobile Subscriber Identity) use AT+CIMI.	0 AT+CIMI 226014052390876 OK
Get the current status of the mobile phone or GSM/GPRS modem. For example, mobile phone activity status (AT+CPAS), mobile network registration status (AT+CREG), radio signal strength (AT+CSQ).	0 AT+CSQ +CSQ: 19,99

6.5 SMS

Settings related to SMS, SMS to e-mail and e-mail to SMS conversion.



6.5.1 Status



Of course, Status shows the current STATE of the e-mail to SMS and SMS to e-mail features, and the addresses of the service centre for the two SIM cards. Here you can only see information, you cannot change the settings.

6.5.2 SMSC Numbers

The phone number of the service centre is very important for sending out SMS messages, no matter if you send them from your cell phone, from the Web interface of SS-16 or from the E-mail-2-SMS program.

The SMSC (SMS centre) is responsible for handling the SMS operations of a mobile network. It routes SMS messages sent from a mobile phones or terminals. When the recipient is not available (out of the coverage area, cell phone switched off), the SMSC will store the SMS messages. Later, when the recipient is available, the SMSC will forward the messages.

Generally the SMS centres are specific to each carrier, so in order to use SMS messaging with your terminal or with SS-16, you must know the correct “address” of the SMSC for the operator of the respective mobile networks.

Normally, the SMSC address is pre-set in the SIM card by the respective mobile wireless network operator, which means **you do not need to make any changes to it**.

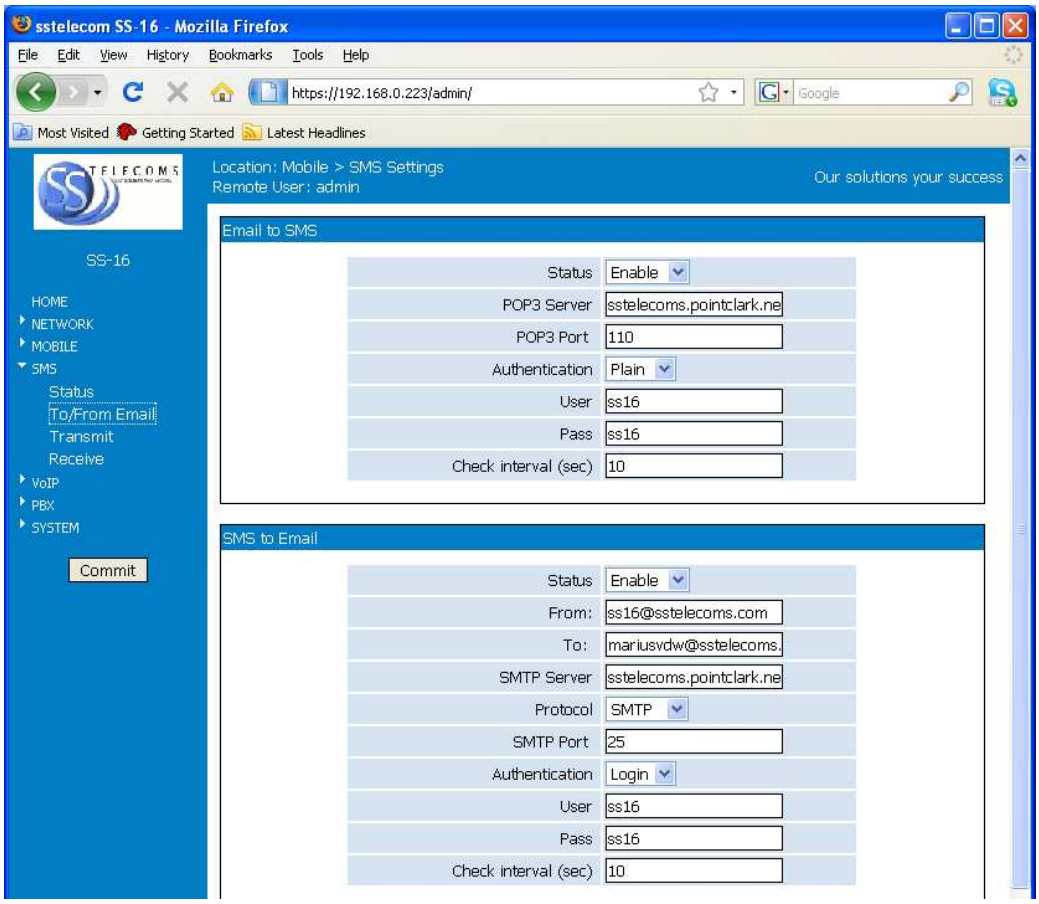
SS-16 gets the SMSC address automatically from each SIM, but you can also change the SC numbers, if there is need to.

To perform this change, you must go to the configuration page **Mobile>At Command** and use the command “**AT+CSCA**” (Service Centre Address) to read or set the address of the service centre through which SMS messages will be sent out:

Mobile Interface: mobile1 Command: +csca=+27831000113	The command **AT+CSCA?** Asks for the current SMSC address, while the command in format **AT+CSCA=number** sets up the new address to the phone number of your choice.

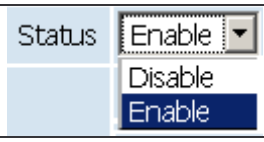
6.5.3 To/From E-Mail

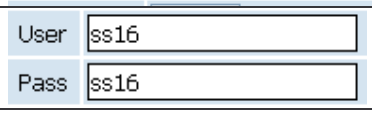
This page controls the E-mail2SMS and SMS2E-mail features.
First, you set up the parameters for the mail gateway application, then you may decide to enable or disable each of the features.
For instance, when SMS to Email is disabled, as shown in the next example, the received messages will be shown in the “Receive” page. If you enable this feature, the SMS messages, as soon as they are received, will be sent out as e-mail messages, and then erased.



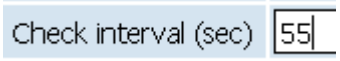
E-mail to SMS
Settings for sending out of e-mail messages as SMS:

You should use a special (dedicated) e-mail account for this.

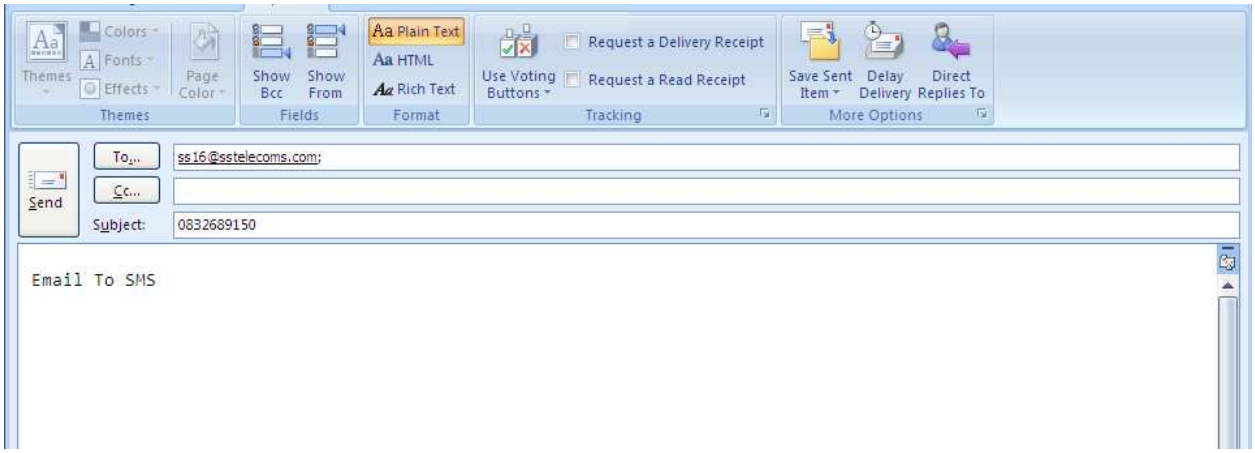
Status: you must Enable this feature, in order to use it. By default it is disabled.	

Settings for the incoming mail server (POP3), the address and the port to be used for the “customer”. Some mail server may use different port numbers, such as 465 or 991, instead of the standard one.	
User name and associated password for the mail account “SMS Outbox”.	

Authentication – method used for authentication, must be set according to the actual requirements of the POP3 mail server.	

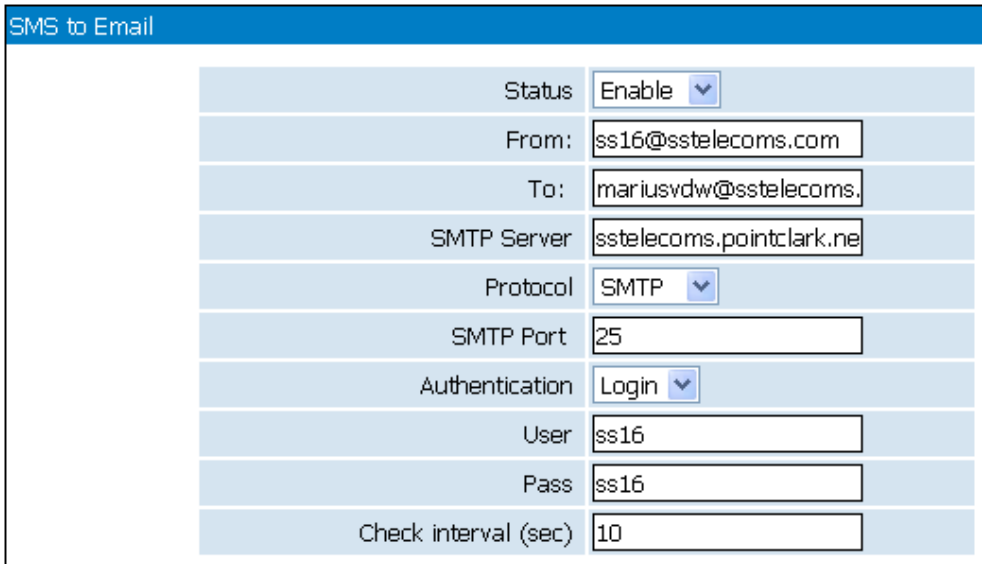
Check interval – you set the value, in seconds, when SS-16 checks for new e-mail messages to send out as SMS	

Using E-mail to SMS feature

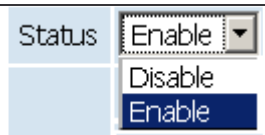


The message must be sent in Plain Tex, not HTML, and will be cut (truncated) at 160 characters. Also, you should use either ASCII or Unicode (UTF-8) encoding. In the “Subject” field, enter the phone numbers(s) as destination for the SMS. You may type a single mobile number, or several phone numbers, using blank space as a delimiter.	

SMS to E-mail
Settings for the SMS to E-mail feature, which converts received SMS into e-mail messages



Status: you must Enable this feature, in order to use it. By default it is also disabled.
When you Enable SMS-to-Email, the received messages will no longer be seen in Receive, because they are deleted as soon as they are sent out as E-mail messages.



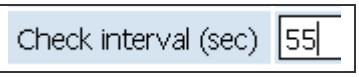
From/To mail settings.
In the FROM field you can enter any address, it is not a "real" e-mail sender. However, you should use a meaningful name.

Settings for the incoming mail server (SMTP), the address and the port to be used for the "customer". Some mail server may use different port numbers, such as 3535, instead of 25 which is the standard one.

User name and associated password for the mail account SMS inbox.

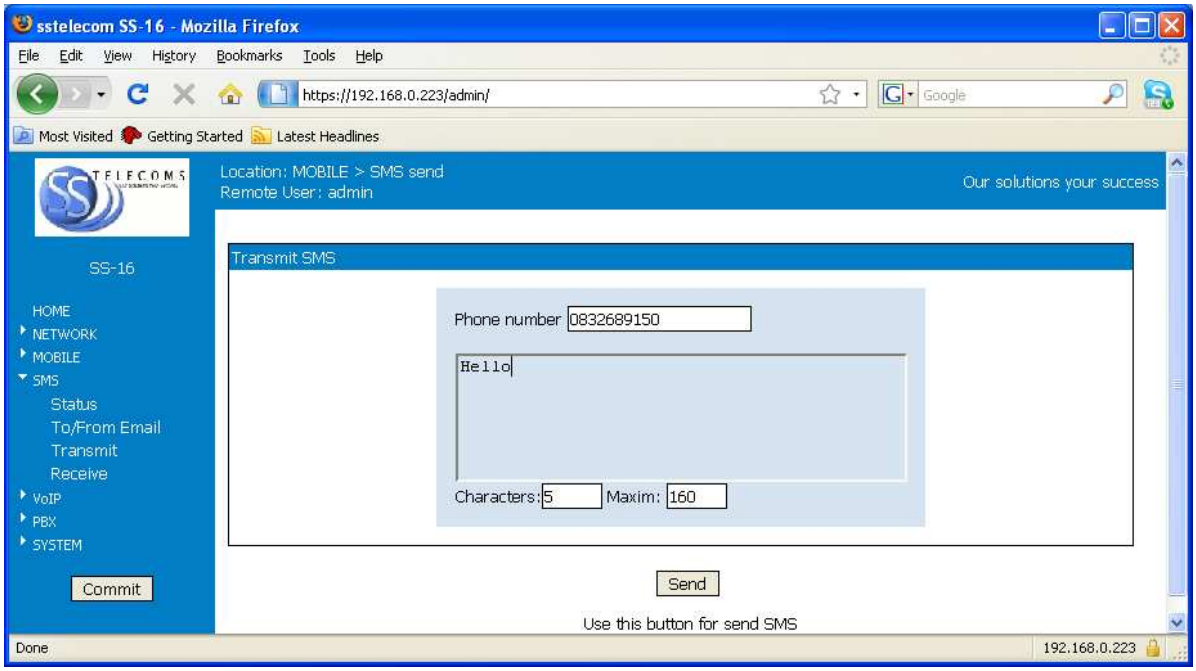
Authentication – method used for authentication, must be set according to the requirements of the SMTP mail server. You may choose either no authentication, plain text password, or with log-in.

Check interval – you set the value, in seconds, when SS-16 IP checks for new SMS to send out as e-mail messages



6.5.4 Transmit

Use this page to send out SMS messages from the Web interface of the SS-16.



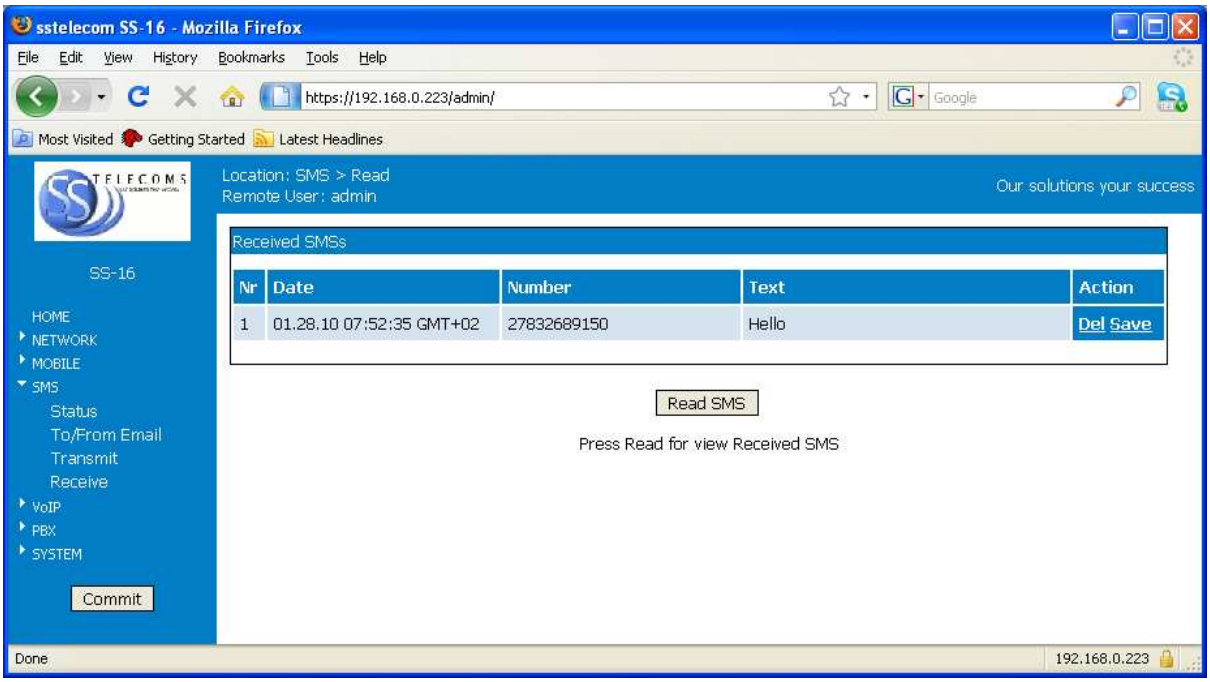
First, type the phone number where the SMS will be sent.
Then, write the text to be sent. You may type it from the keyboard, or use Copy and Paste to import text from other applications (mail, word processing, etc)
For ASCII characters, the maximum length is the standard one for SMS, **160 characters**.
The window “Characters” show you how may characters you have already typed, 91 out of 160 in the above example.

If you continue to type letters after you have reached the limit of 160 characters, you will see this warning, telling you that the outgoing SMS will be truncated at 160 chars.

Finally, you should see a confirmation message on the screen:

6.5.5 Receive

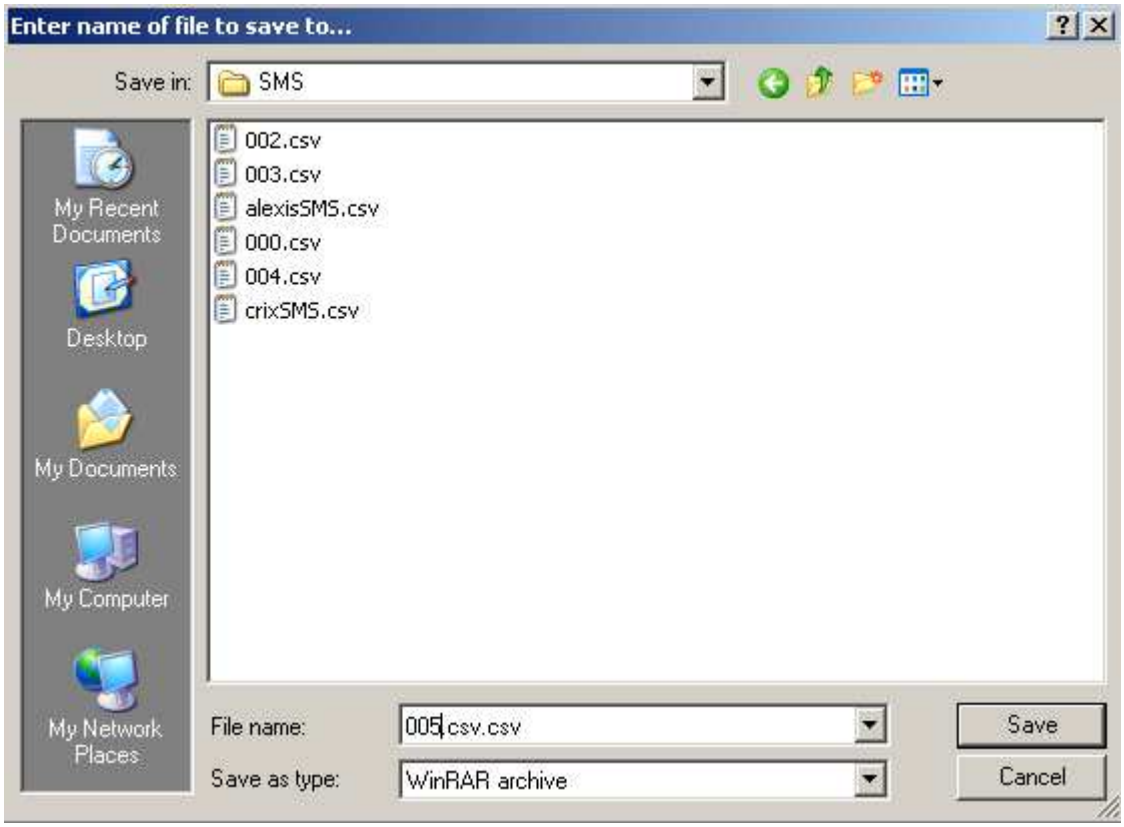
Here you can see the SMS messages received.



Click the “read” button to refresh the list of messages.

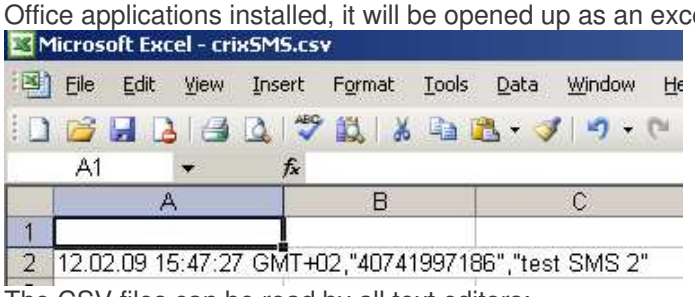
The received messages are shown in this page only when the feature “SMS to E-mail” is disabled. If it is enabled, the received SMS are sent out as e-mail, and then erased, so you won’t be able to see them in the Receive list!

The last column, the field “Action”, allows you to individually delete or save the SMS messages. Use “Save” to save the respective SMS in a folder of your choice, in format csv.	

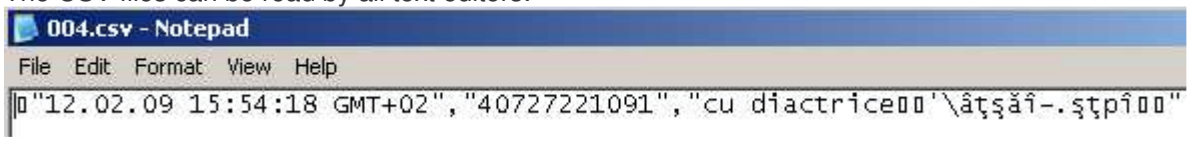


Afterwards, you can go to the respective folder and open up the saved SMS.

The “Comma Separated Values” format is understood by many applications, for instance if you have Office applications installed, it will be opened up as an excel spreadsheet:

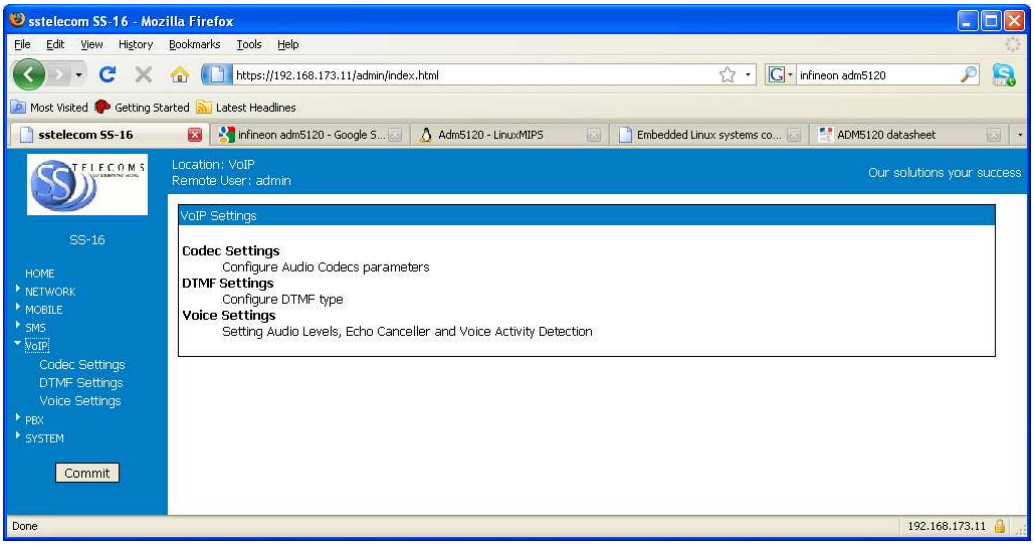


The CSV files can be read by all text editors:



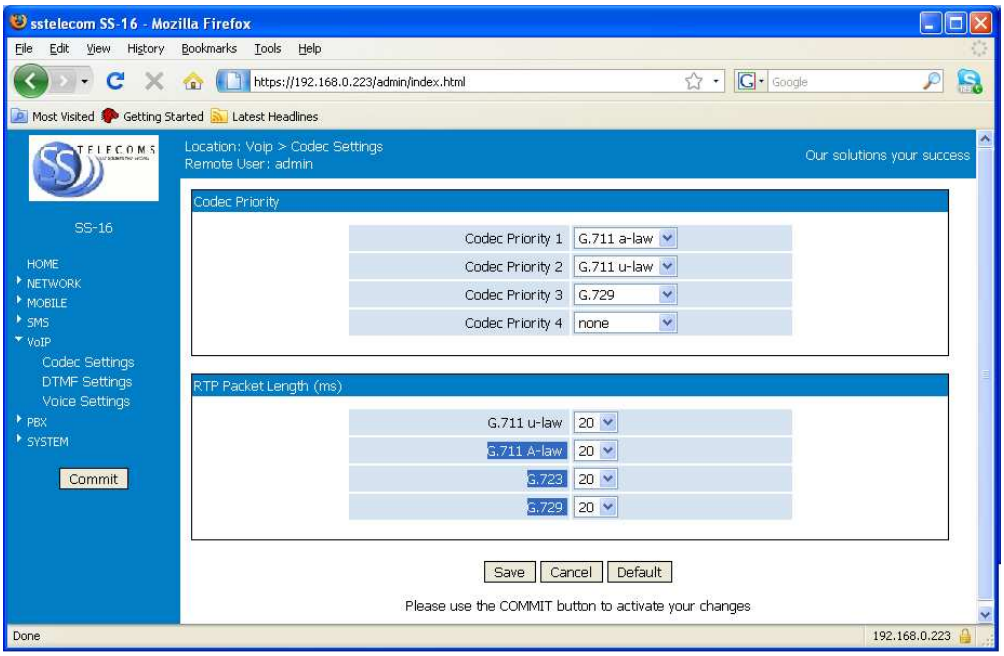
6.6 VOIP

Allows one to configure the settings for Voice over IP aspect of the calls, related to the codecs used and transmission of DTMF signalling.

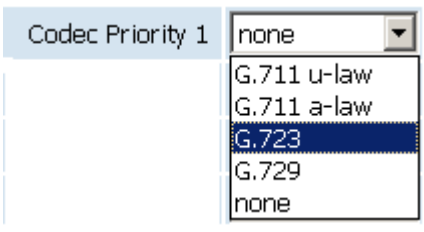


6.6.1 Codec Settings

Settings for the codecs used by the Voice over IP calls:



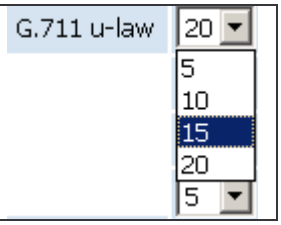
Codec priority

SS-16 VOIP-2-GSM supports several audio codecs, currently G.711 (both A-law and u-law), 723 and 729. There are four boxes, with decreasing priorities, from 1 down to 4. For each one, one may assign a codec, or select “none”. Generally one should enter the codec G.711 on the first place, either with A-Law for Europe or u-law for North America, followed by G.729. G.711 provides good voice quality and	

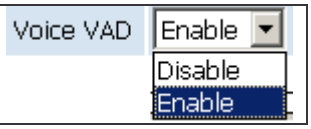
works well with in-band DTMF.	

If the IP-PBX or the SIP-telephones that will be connected with SS-16 VOIP-2-GSM make use of different codecs, one should set the “Codec Propriety” field accordingly.

RTP Packet Length

It lets you establish the value for packetization time (in milliseconds) for each codec. It is usually set to 20, but it may be as long as 30 msec or as short as 5 msec. Increasing the packetization time reduces the packetization overhead, but increases the latency.	

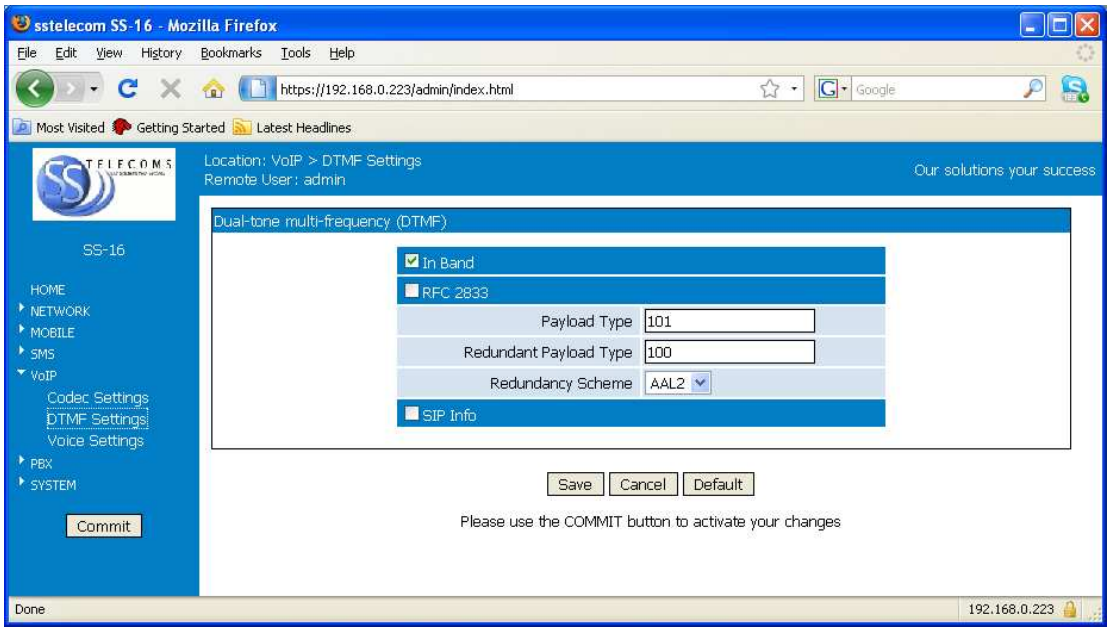
Voice VAD

Enables or disables the "Voice Activity Detection", by default this feature is disabled.	

Typical voice conversations can contain up to 35 to 50 percent silence. On VoIP networks, both conversation and silence is packetized. If VAD is enabled, the equipment sends out RTP packets only when voice is detected, thus decreasing bandwidth by 30-50 %. This way the SS-16 VOIP-2-GSM will send voice packets only when it has voice input. When it detects silence on RTP, it will send a “silence packet” to the other side which uses less bandwidth and allows to be interpreted for generating comfort noise.

6.6.2 DTMF Settings

Settings for sending out DTMF in VoIP calls:



This configuration area establishes the means of sending out DTMF into the VoIP calls. There are three methods available, which may be used at the same time. Transmission of DTMF signalling can be done in band, out of band in RTP packets, or sending out INFO messages (as digital telephone-events).

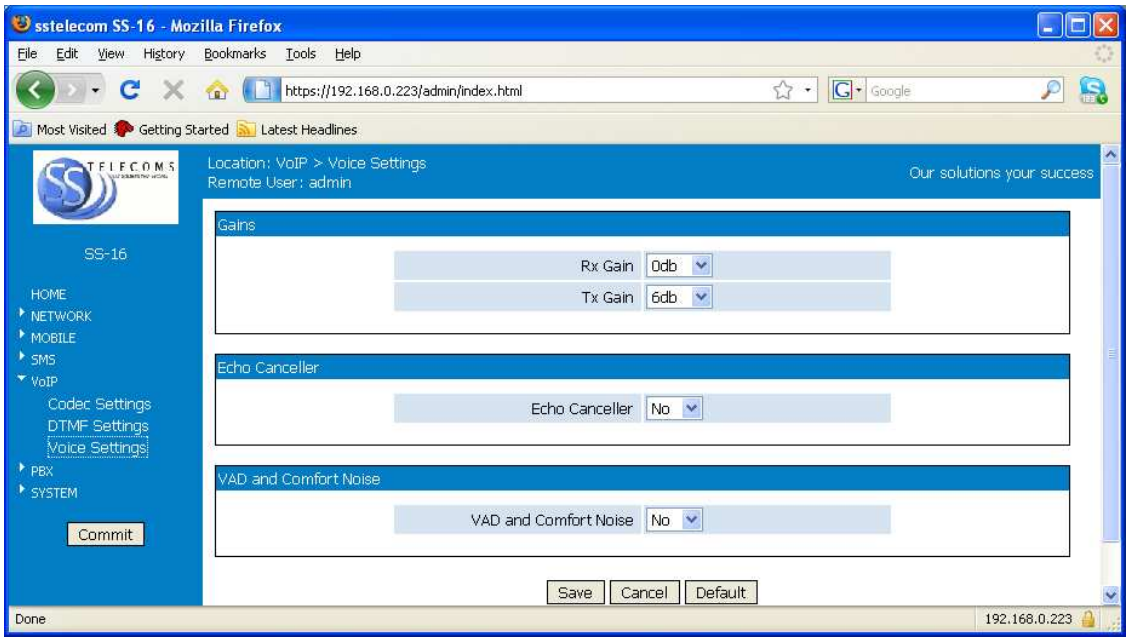
Under the SIP protocol, the INFO method would be used for carrying of mid-call signalling information along the session signalling path (for example to send the DTMF digits generated during a SIP session). Check the box “SIP Info” for this way of transmitting DTMF.

When checking this “RFC 2833” options, also complete the next Parameters fields to the right

- payload type for DTMF in RTP (according to RFC 2833)
- redundancy payload type
- redundancy scheme: either IETF or =AAL2.

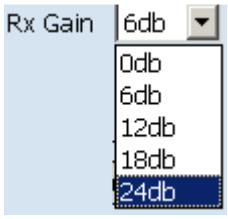
5.5.3 Voice Settings

Adjust the gain (amplification) on the VoIP side of the equipment:



In the same way as for the mobile modules, but for the Voice over IP section, here you can adjust the audio gain for reception and transmission.

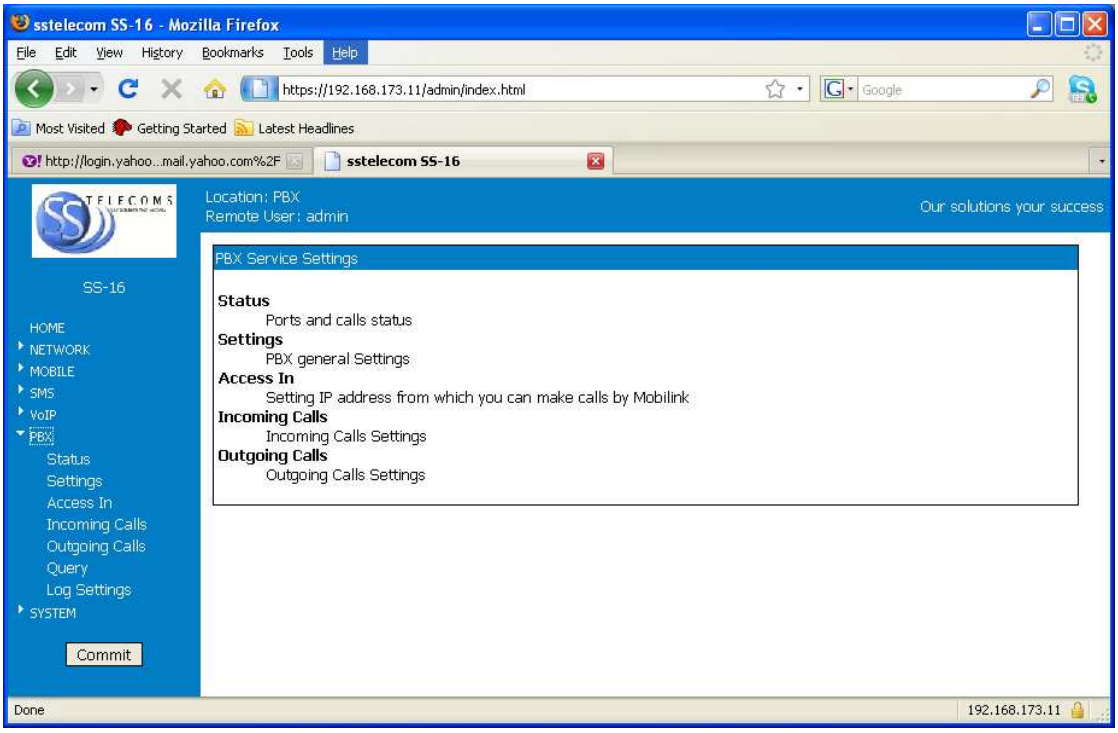
Choose Rx or Tx, and form the drop list, select the amplification you want, from zero dB up to 24 db, in 6 dB increments.



Definition of RX and TX
In this page for VoIP voice Settings, RX is the voice path going out from VoIP towards the mobile network, while TX is the reciprocal voice path, which comes from the mobile network towards VoIP. The meaning of Tx and respectively Rx are reversed compared to the page “Mobile Settings”!

6.7 PBX

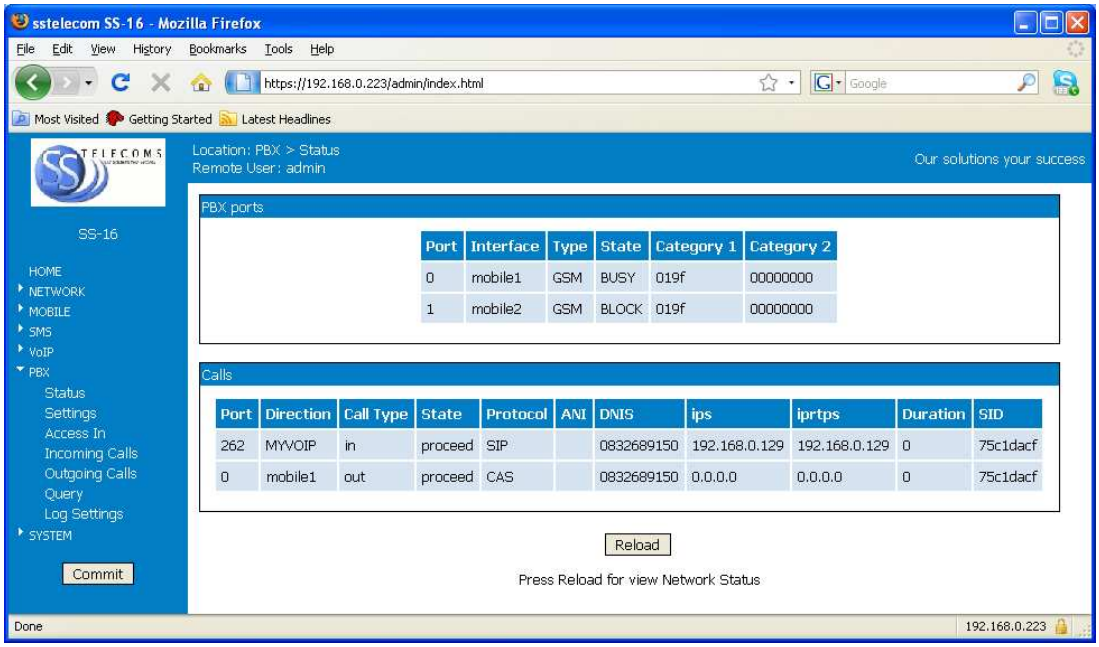
Controls the settings related to the phone exchange features of the SS-16 VOIP-2-GSM.



This application runs on the SS-16 VOIP-2-GSM equipment and performs all the tasks of call-control (management, routing).

6.5.1 Status

Displays the current state of the ports of the embedded “phone exchange” as well as the status of the calls:



In the stand-by status, both ports are “free” and no info about calls is displayed, since there are currently no conversations going through the SS-16 VOIP-2-GSM.

PBX ports

Shows the state of the two mobile (GSM) interfaces of the equipment, from the point of view of the call control application. They may be FREE (available), BLOCK (blocked for calls) or BUSY (in a call).
Note: The display "BLOCK" means that NOW the respective Mobile interface is not available, but it may have several different reasons:

State
BUSY
BLOCK

State
BUSY
FREE

State
FREE
BUSY

The module was disabled from the Web interface, it does not have a good SIM, or simply the device has been rebooted and the respective module is in the initialization or searching phase, and after a delay it may become again available, such as shown below!

Port	Interface	Type	State	Category 1	Cat
0	mobile1	GSM	BLOCK	019f	000
1	mobile2	GSM	BLOCK	019f	000

Calls

Displays info about the calls routed through the SS-16 VOIP-2-GSM.

Calls

Port	Direction	Call Type	State	Protocol	ANI	DNIS	ips	iprtps	Duration	SID
262	MYVOIP	in	proceed	SIP		0832689150	192.168.0.129	192.168.0.129	0	75c1dacf
0	mobile1	out	proceed	CAS		0832689150	0.0.0.0	0.0.0.0	0	75c1dacf

The example above shows a single VoIP call coming from a Soft-phone application running on computer with the IP address "192.168.0.129" in the local network and getting out of the SS-16 VOIP-2-GSM through the GSM interface Mobile1. The mobile number called is 0832….

Ports – 1, 2 are the two Mobile ports, while 261, 262 are VoIP ports

Direction – the calls may belong to one of the directions MYVOIP, SIP, mobile1, mobile2

Type – type of the call, it may be "in" for incoming or "out" for outgoing.

State – current status of the call, "alerting", "proceed" or "connected"

Protocol – SIP for VoIP, CAS for the mobile networks, or "Unknown"

ANI – identity of the calling party, received by the SIP user. Generally, Automatic Number Identification is a service that provides the receiver of a telephone call with the number of the calling phone.

DNIS – destination number, dialled by the SIP user. Dialled number identification service, also used for the routing of calls based upon the phone number of the destination.

Ips – IP address used for Signalling, SIP, such as 192.168.0.129 in the above example.

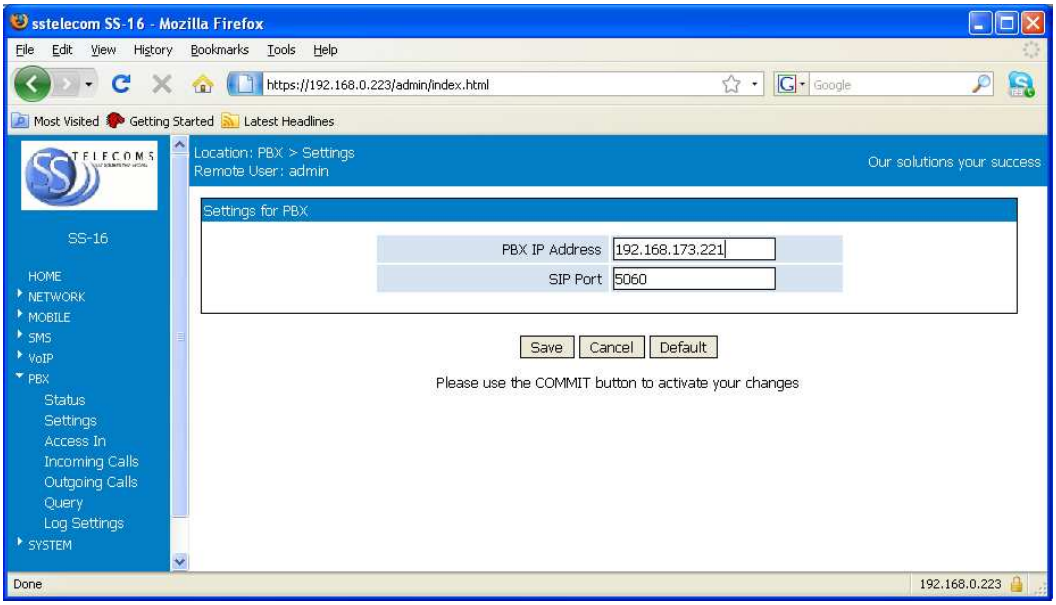
Iprtps – IP address for RTP may be the same as the signalling address, or it may be two different addresses. There are examples of both cases.

Duration - total duration of the call, in seconds, starting from the moment the called party answers.

SID –Session ID, each session for every call has a unique ID

6.5.2 Settings

Configure the settings for the IP phone exchange that will work with the SS-16.

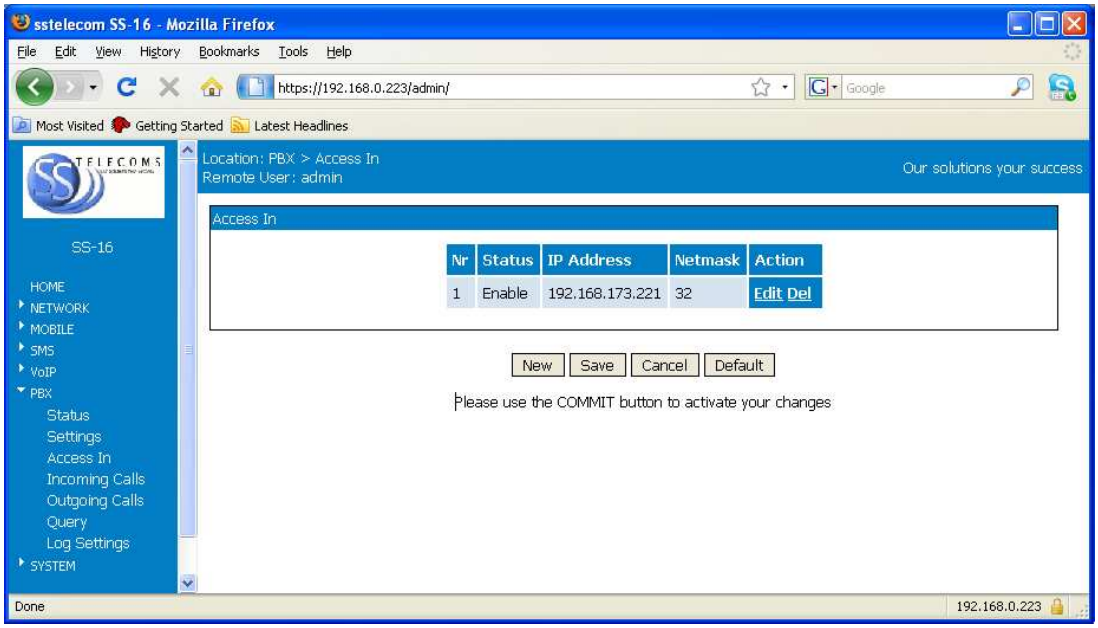


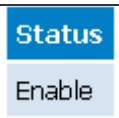
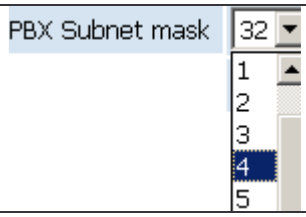
Select the parameters for the IP phone exchange (PABX) where the SS Telecoms VoIP-2-mobile interface is connected: **IP address**, subnet mask, and SIP port.

SIP Port – number of the port used for SIP calls. Here the port used for SIP is the standard 5060, but the user may change this value if the SIP PBX operates differently.

6.5.3 Access In

SIP Users / IP addresses that will be able to use the SS-16 VOIP-2-GSM equipment for out going calls:



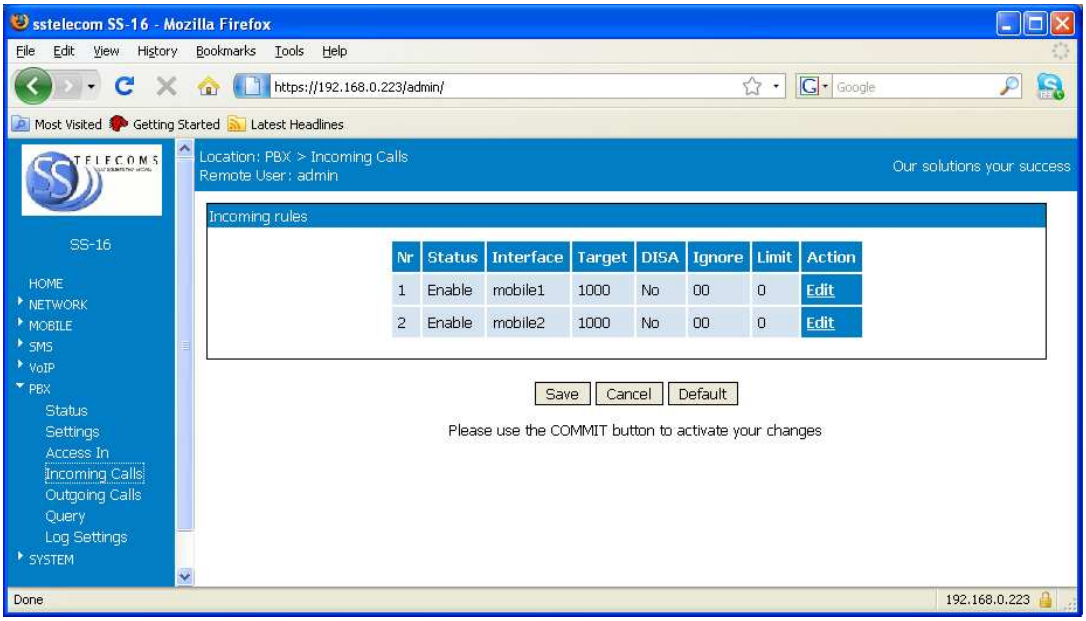
Status – Enable or disables rule.	
IP Address – The allowed IP address range. In this example one would use a Subnet mask of 32. This means that the device with this IP address will have access to make outgoing calls. Normally this would be set to the IP address of the PABX also configured under PBX settings.	
Subnet mask – establishes the sub-network of the PABX.	

Examples

IPAddress	Subnet Mask	Note
0.0.0.0	0	Calls will be accepted from any IP Address.
192.0.0.0	8	IP Address between 192.0.0.1 to 192.254.254.254 allowed.
192.168.0.0	16	IP Address between 192.168.0.1 to 192.168.254.254 allowed.
192.168.173.0	24	IP Address between 192.168.173.1 to 192.168.173.254 allowed.
192.168.173.221	32	Call will only be accepted from 192.168.173.221

6.5.4 Incoming calls

Settings for the incoming calls, that means calls which come in through the mobile interfaces of the equipment:



Here there are only two rows that can be edited - it cannot be deleted or more rows added to it. Each row sets the rules for one GSM module – interfaces Mobile 1 and respectively Mobile 2:

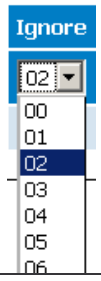
Define the rules for calls coming in from the mobile networks.
For each of the two Mobile interfaces one can choose Enable or Disable and select the Target (destination) and Action to be taken. First click Edit to start editing the specific row, then Exit and save the settings:

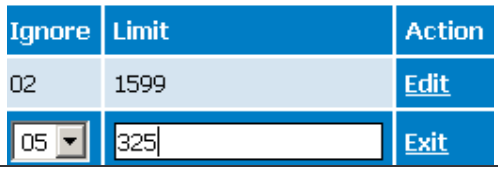
Nr	Status	Interface	Target	DISA	Ignore	Limit	Action
1	Enable	mobile1	1000	No	00	0	Edit
2	Enable	mobile2	1000	No	00	0	Edit

Target – the number of the local extension where the calls will be routed which are coming in through the respective Mobile interface (1000, etc in the examples used).
DISA – select **No** (default) or **Yes** from the drop list.

If choosing YES, the value in the field Target doesn't matter any longer, the caller will receive a DISA tone allowing him to dial the number of the local extension he wants to reach. See below the routing of such a call, coming in through interface Mobile1 and being connected to destination by means of DISA dialling:

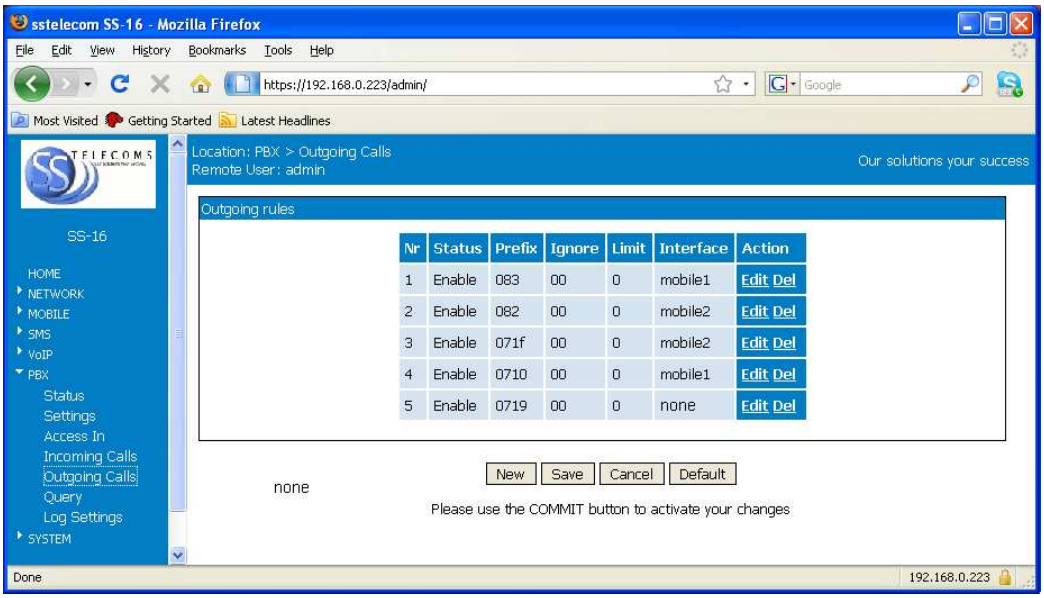
Ignore
You may set SS16 to ignore the incoming Caller Identity. From the drop list, select the number of digits to be ignored, from "00" (no digits are ignored) up to "15" .



Limit Limit for incoming calls in seconds.	

6.5.5 Outgoing calls

Establishes the rules for outgoing calls, which are calls coming from SIP and directed to mobile network.



One can define a table with several rules for calls coming in from SIP and going out through the mobile interfaces.

Each rule may be individually enabled or disabled. The program analyses the numbering (the prefix) of the phone number called and routes them accordingly through one of the mobile interfaces.
If one of the mobile ports is disabled, the respective GSM module will NOT be used for outgoing calls.

Use the **New** button to create a new rule, **Edit** to fill the parameters for it, **Exit** to get out of the respective row, and **Save** to save the rules created.

The button **Del** is to erase an existing rule from the table.

Example:

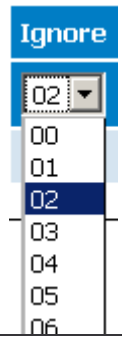
If the call has the prefix 0711, it will go out through the interface Mobile2, if it has the prefix 0710, it will go out through the interface Mobile1, and so on.

The prefix “071f” means 071 followed by any digit (no matter which ones), one must use this format for compatibility – all the prefixes in the table must have the same number of digits.
The value “any” in the field “Interface” means that the respective call can be routed via any of the two modules.

To barr or block a prefix select none as the Interface destination.

Ignore:
You can set the device to ignore between 0 and 5 digits on the out going called phone number.

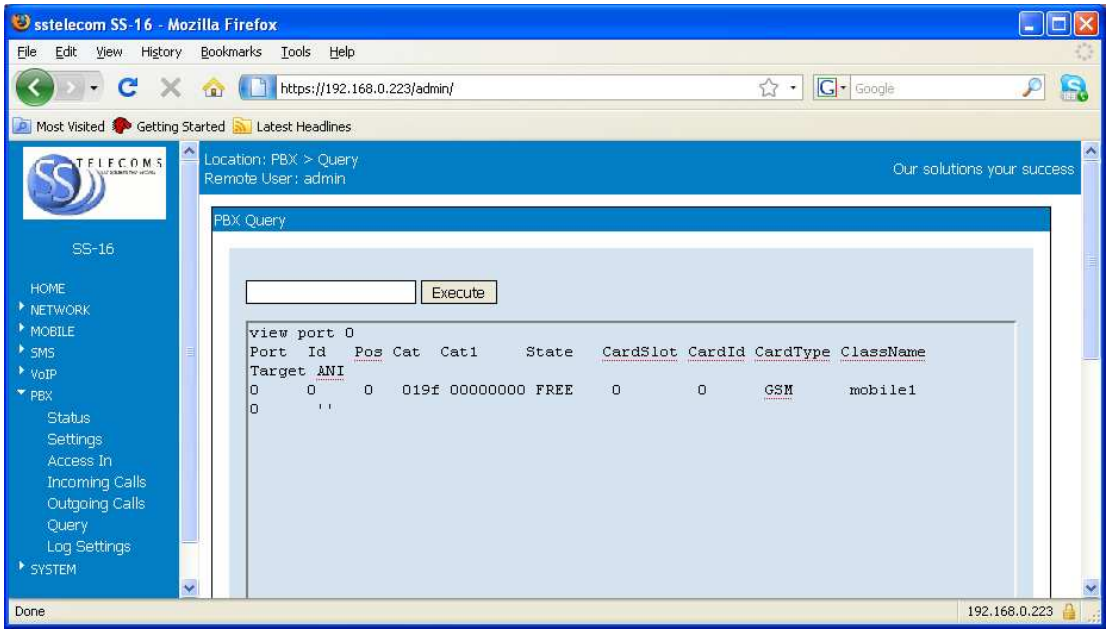
Example:
If the ignore field is set to 01 and the clients calls 00832689150 the first number will be ignored and 0832689150 will be dialled.



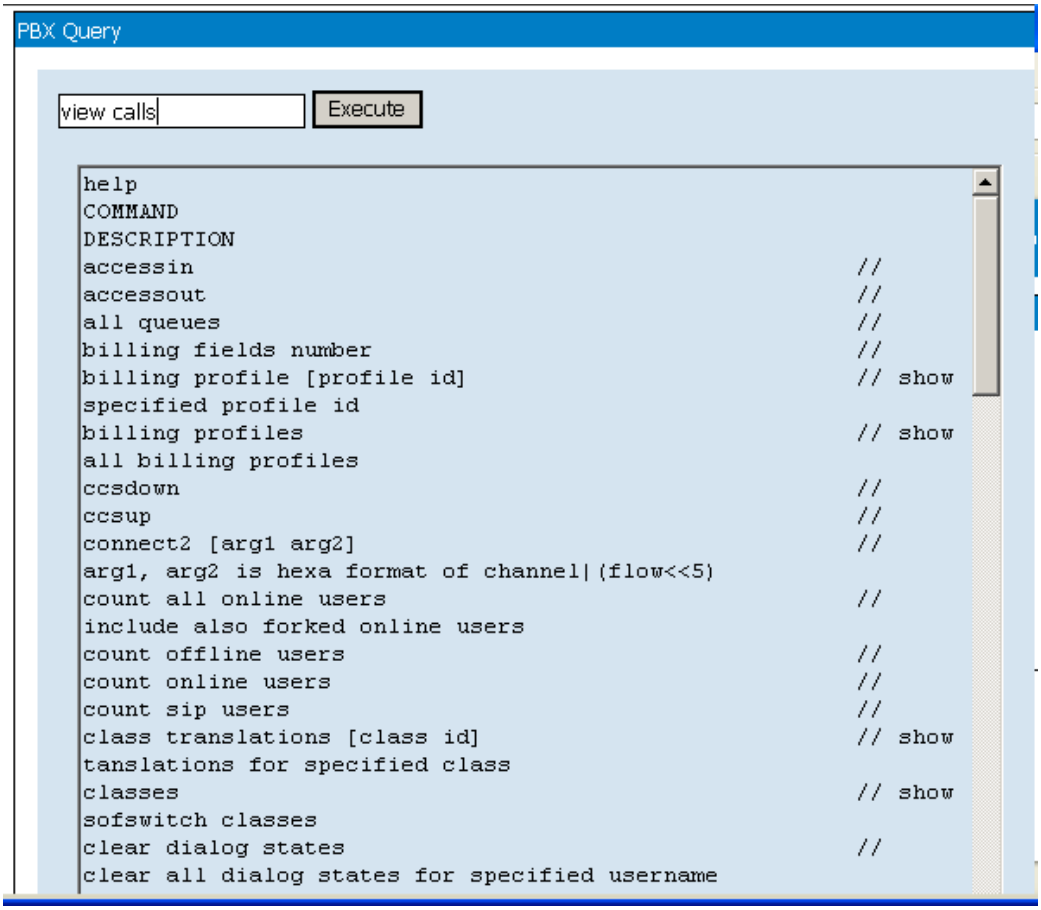
Limit
Limit for out going calls in seconds.

6.5.6 PBX Querry

Allows the sending of text commands to the phone exchange application which runs on the SS-16 VOIP-2-GSM equipment.

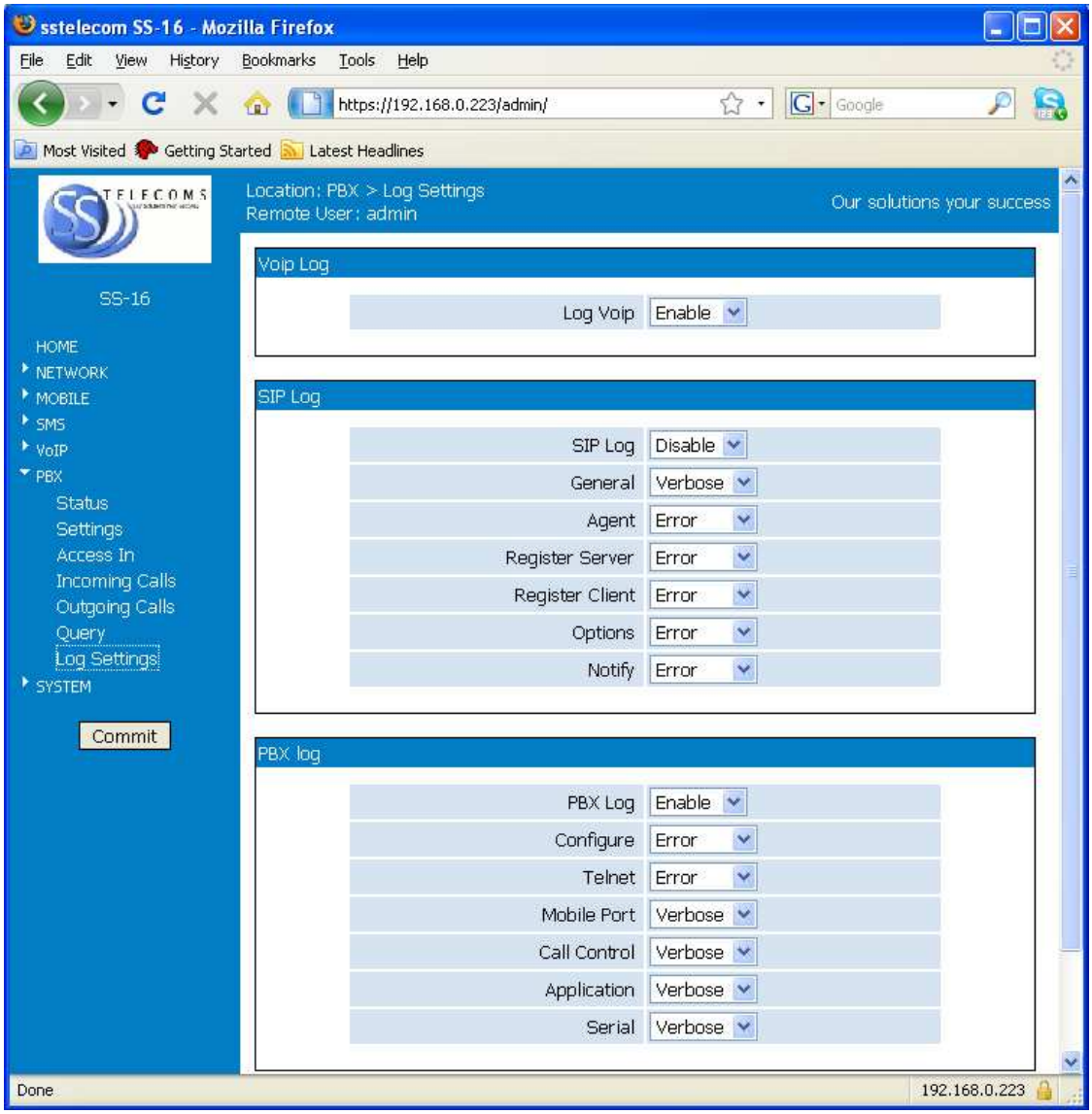


The command “help” shows you all available commands:

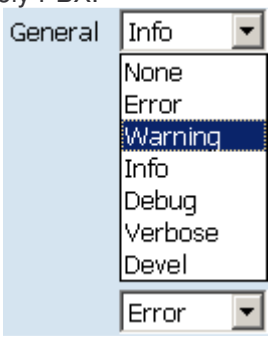


6.5.7 Log Settings

Here one can set up how much information will be stored for each section of the log.

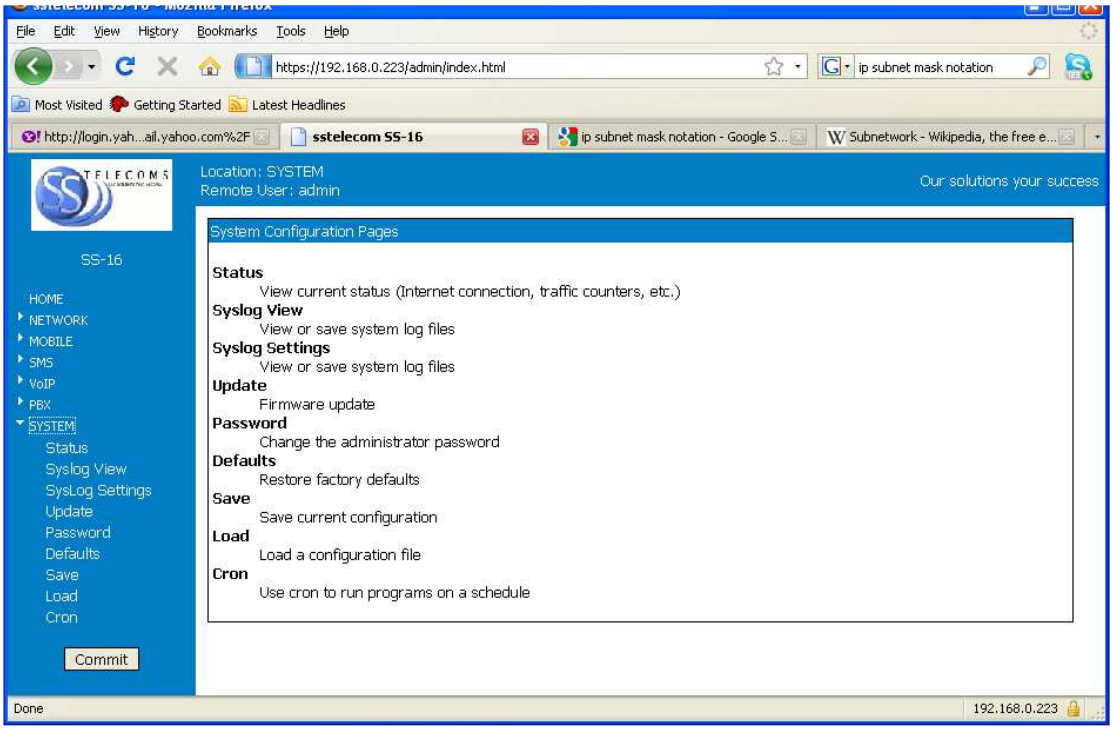


The sections of the log are: Voice over IP, SIP and respectively PBX. In each section there are one or several items, and for each of these items one may choose the level of details of the log form the drop list that shows up: None, Error, Info, Warning, Debug, Devel, Verbose and so on. “None” or “Disable” means no log at all is created for the respective element, “Error” is the minimal level (only errors will be logged), while “Devel” is the maximum details setting – everything is logged, for the usage of developers.



6.8 SYSTEM

It shows the current state of the SS-16 equipment and the logs also allows performing several system-related operations.

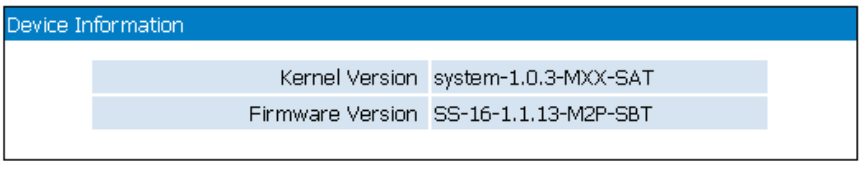


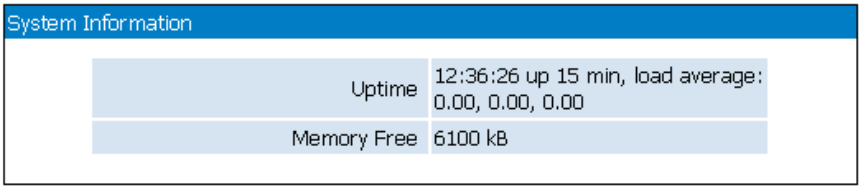
The system configuration pages include these settings:

- Status, displays the current status of the SS-16 VOIP-2-GSM equipment
- Syslog View, shows the log file and allows you to save it
- Syslog Settings, settings for the remote log
- Update, you may perform firmware update or upgrade
- Password, allows modification of the password
- Defaults, restores the factory default settings
- Save, saves the current configuration of SS-16 VOIP-2-GSM
- Load, loads a configuration previously stored.
- Cron, Used to run programs on a schedule.

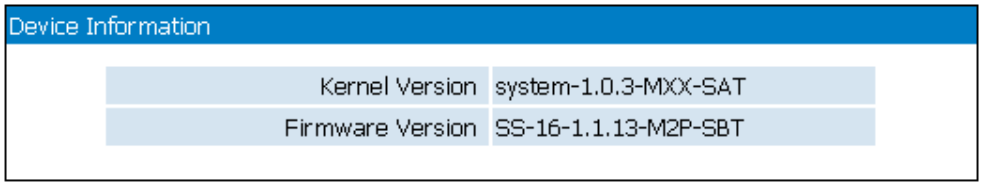
6.8.1 Status

Shows the status of the SS-16 VOIP-2-GSM system. In some firmware versions, System Status and other special Admin functions may be protected by additional passwords.





This page displays information about the following items:



- Firmware version: version of the application firmware running on SS-16 VOIP-2-GSM: you may see “Firmware version is 1.1.13” .The firmware can be easily updated (a new version) or upgraded (new software features added)

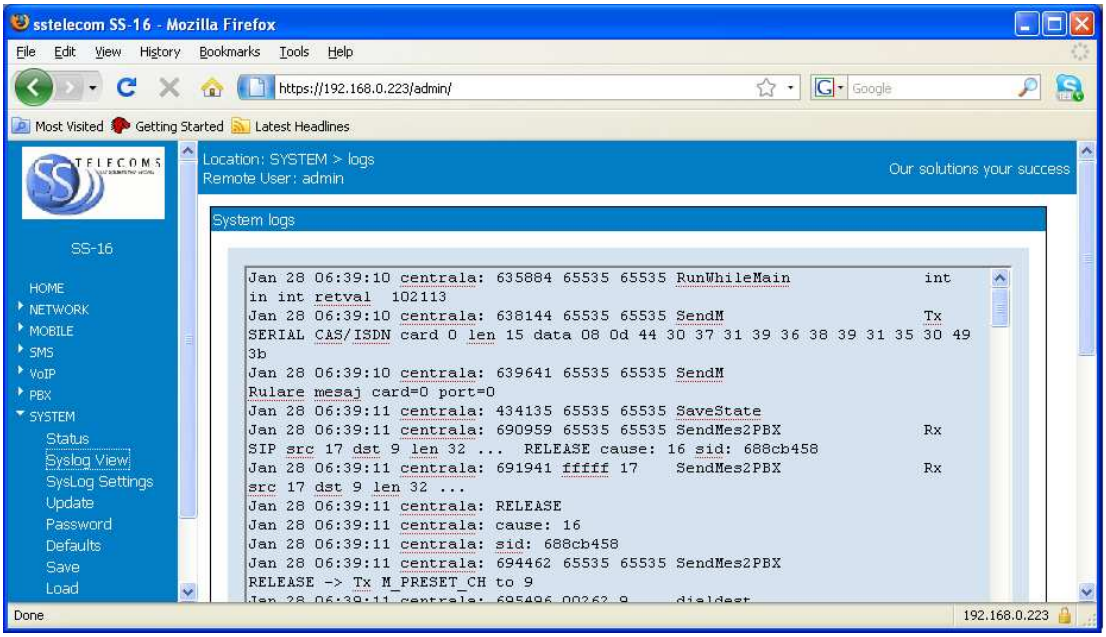
- Kernel version (Operating system version), such as system-1.0.3 in the example above. The application firmware and the operating system (root or kernel) may be updated separately, according to the requirements of the users. The operating system memory is write-protected, thus in case of problems, accidental erasure or power failure during updating, you still have a functional “kernel” allowing you to load a new software image;

System uptime: time since the last restart of the SS-16 VOIP-2-GSM, both in full format (hours: minutes: seconds) and short format (up 1 minute) “System uptime: 11:41:59 up 4:50”;

- Load average: “0.06, 0.04, 0.01” or “load average: 1.50, 1.67, 1.71 “ - info about system loading (maximum value / average value / number of active processes)

6.8.2 Syslog View

Shows the system log:


sstelecom SS-16 - Mozilla Firefox
File Edit View History Bookmarks Tools Help
https://192.168.0.223/admin/
Most Visited Getting Started Latest Headlines
Location: SYSTEM > logs
Remote User: admin
Our solutions your success
SS-16
HOME
NETWORK
MOBILE
SMS
VoIP
PBX
SYSTEM
Status
Syslog View
SysLog Settings
Update
Password
Defaults
Save
Load
System logs
Jan 28 06:39:10 centrala: 635884 65535 65535 RunWhileMain int
in int retval 102113
Jan 28 06:39:10 centrala: 638144 65535 65535 SendM Tx
SERIAL CAS/ISDN card 0 len 15 data 08 0d 44 30 37 31 39 36 38 39 31 35 30 49
3b
Jan 28 06:39:10 centrala: 639641 65535 65535 SendM
Rulare mesaj card=0 port=0
Jan 28 06:39:11 centrala: 434135 65535 65535 SaveState
Jan 28 06:39:11 centrala: 690959 65535 65535 SendMes2PBX Rx
SIP src 17 dst 9 len 32 ... RELEASE cause: 16 sid: 688cb458
Jan 28 06:39:11 centrala: 691941 fffff 17 SendMes2PBX Rx
src 17 dst 9 len 32 ...
Jan 28 06:39:11 centrala: RELEASE
Jan 28 06:39:11 centrala: cause: 16
Jan 28 06:39:11 centrala: sid: 688cb458
Jan 28 06:39:11 centrala: 694462 65535 65535 SendMes2PBX
RELEASE -> Tx M_PRESET_CH to 9
Done
192.168.0.223

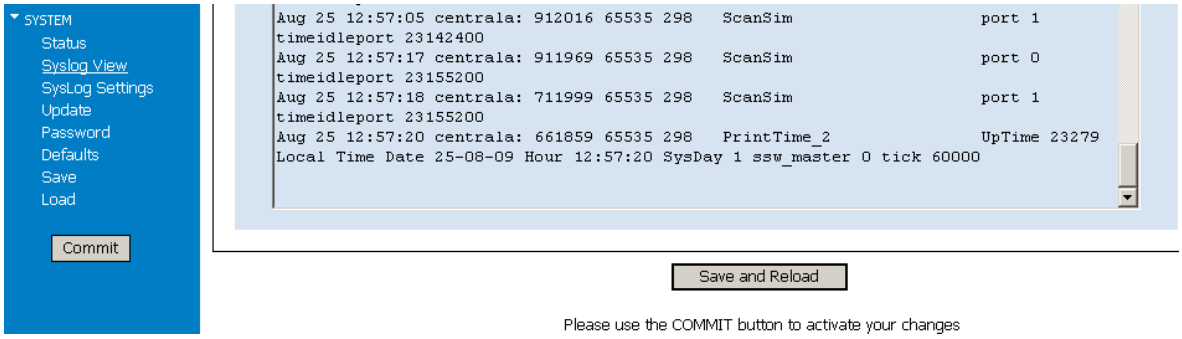
The log is a place where all the applications running on the SS-16 VOIP-2-GSM store their output messages: status reports, confirmation or error messages, activity, etc.

It is a plain text (thus easy to interpret) record of actions taken by the software as it runs on SS-16 VOIP-2-GSM: changes made, devices and drivers detected, files added or deleted, communication with the modem, network settings, etc.

Correspondingly, records of the kernel or of the user info, debug messages, info or warnings can be seen in the log:

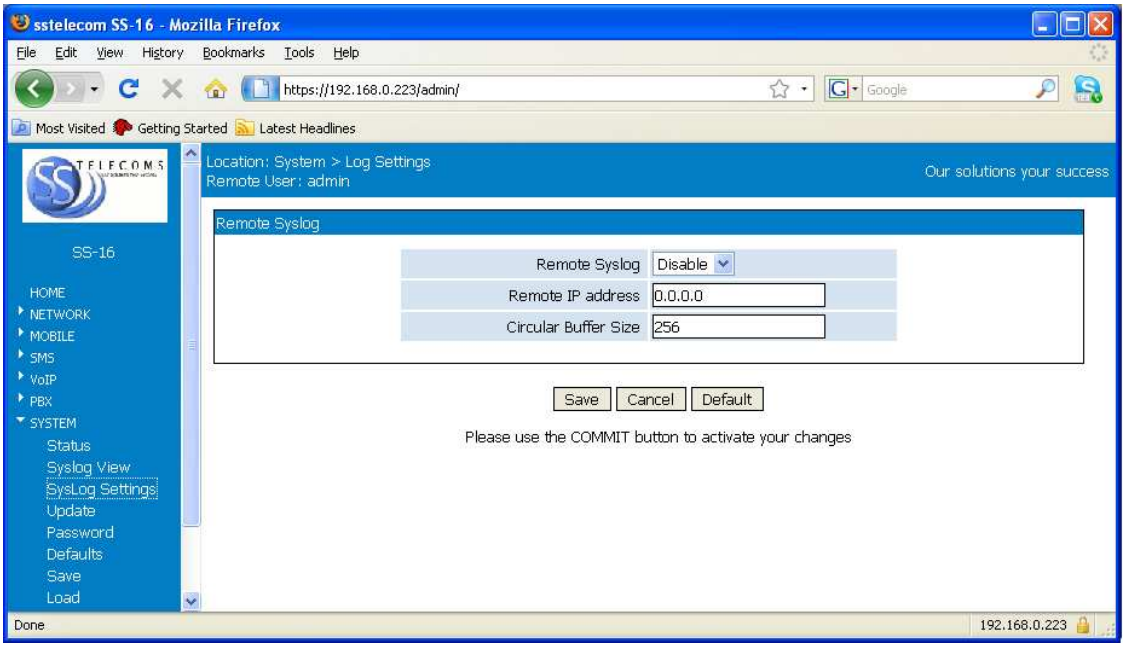
The log is very helpful when troubleshooting. It may be saved on a PC (simply use "Save As" option of your browser) for further examination.

Use the button "Save and reload" at the bottom of the Syslog window to save the current log and refresh the display:


SYSTEM
Status
Syslog View
SysLog Settings
Update
Password
Defaults
Save
Load
Commit
Aug 25 12:57:05 centrala: 912016 65535 298 ScanSim port 1
timeidleport 23142400
Aug 25 12:57:17 centrala: 911969 65535 298 ScanSim port 0
timeidleport 23155200
Aug 25 12:57:18 centrala: 711999 65535 298 ScanSim port 1
timeidleport 23155200
Aug 25 12:57:20 centrala: 661859 65535 298 PrintTime_2 UpTime 23279
Local Time Date 25-08-09 Hour 12:57:20 SysDay 1 ssw_master 0 tick 60000
Save and Reload
Please use the COMMIT button to activate your changes

6.8.3 Syslog Settings

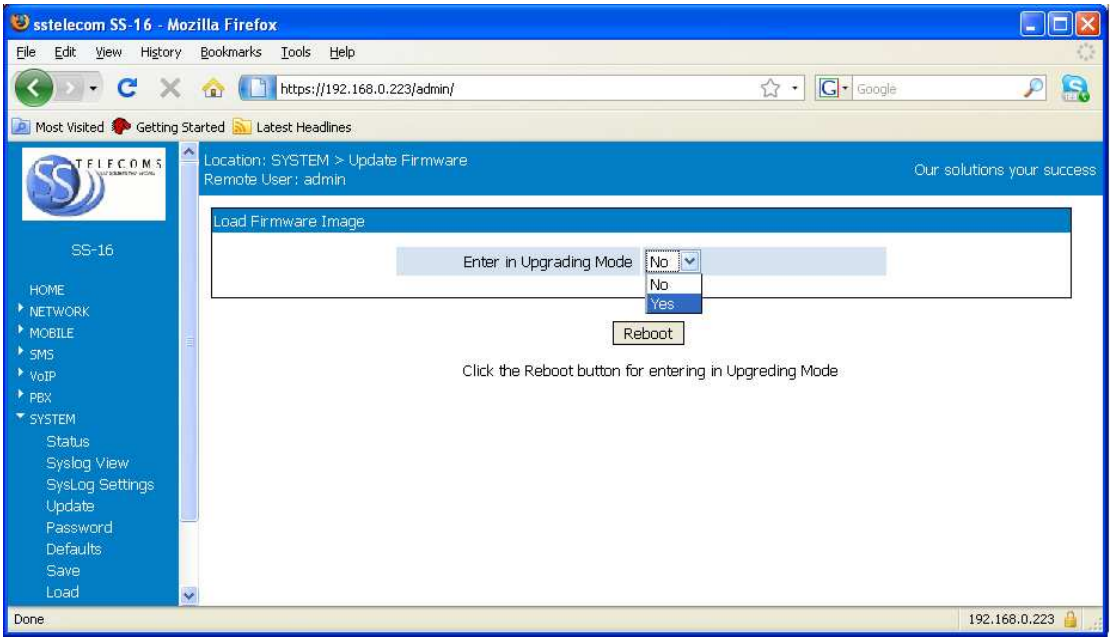
Settings for the remote system log:



By default, the remote log is disabled. If selecting Enable and enter a valid IP, instead of showing the log file locally, SS-16 IP will send it to the remote machine.
The user can also establish the size of the circular buffer that temporary stores the log.

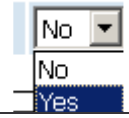
6.8.4 Update

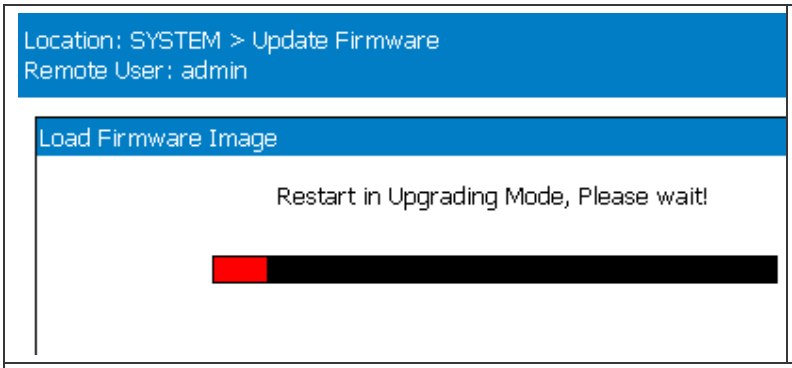
Allows one to perform an update or upgrade of the firmware running on the SS-16 VOIP-2-GSM.



The webpage displays a single field, "**Enter in Upgrading Mode No/Yes**".

As you can see, the default is No, to prevent accidentally going to the mode of operation "Upgrade". If you do want to perform a software upgrade, select YES from the drop list, then click the button "Reboot".





The screen will display the message "Restart in Upgrading Mode, Please wait!" and below it you will see a progress bar:

The SS-16 will shut down and restart in Upgrading Mode.

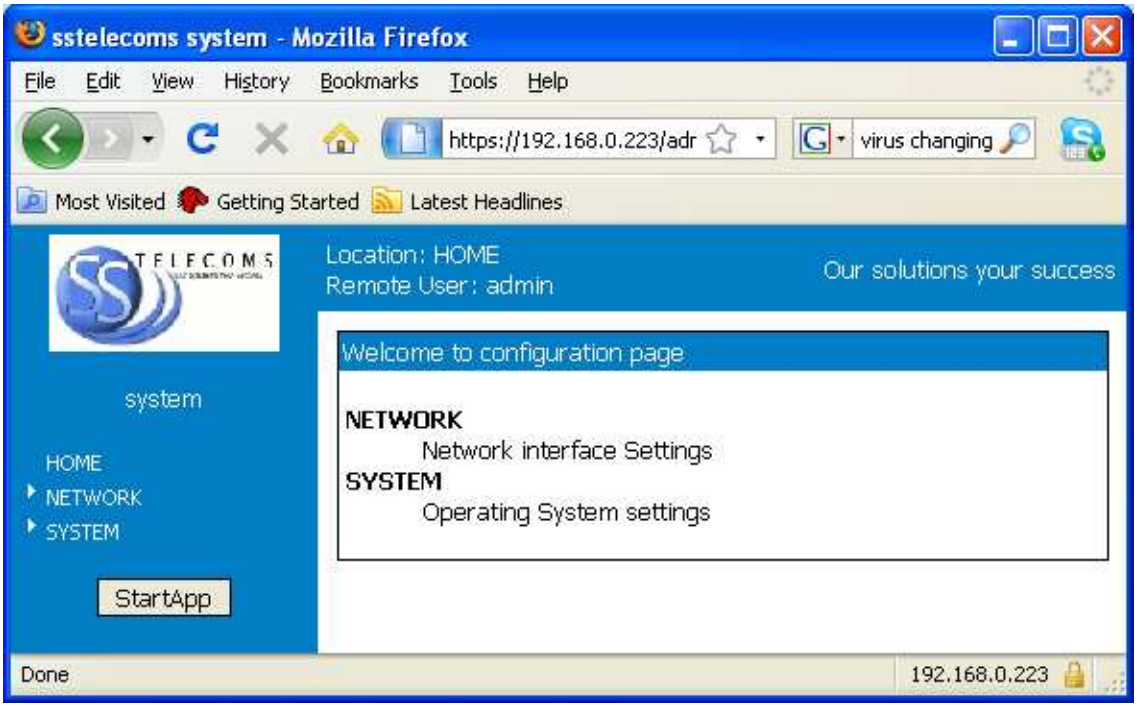
This special mode of operation frees up the memory of the equipment for the new firmware, allowing you to perform software upgrade easily and safely.

The system displays information about each step of the updating process, as shown here:

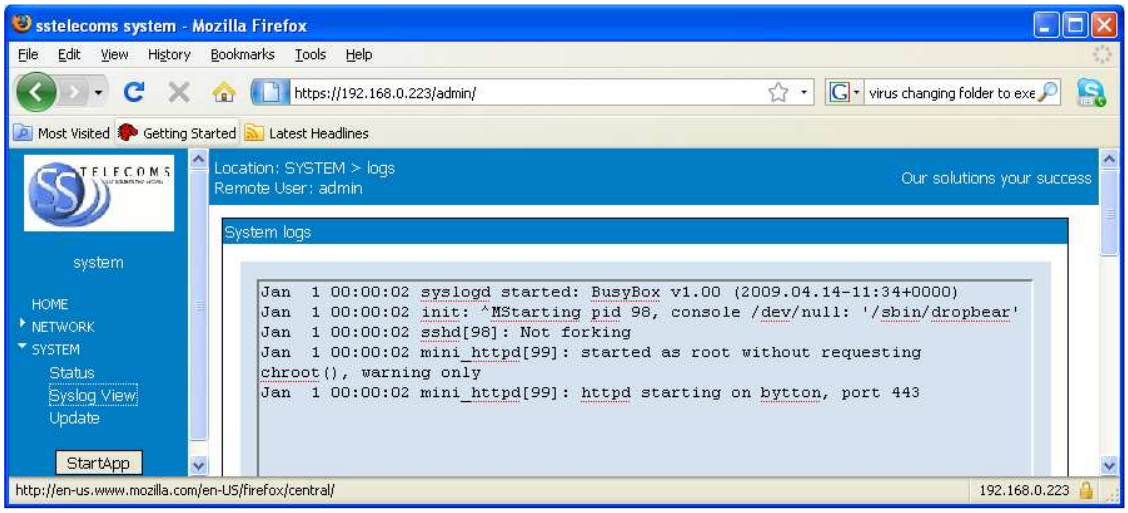
During the updating process, SS-16 is not available for routing VoIP and mobile calls.

Also, if an error occurs during software update and the image file gets corrupted, the equipment will restart in System mode.

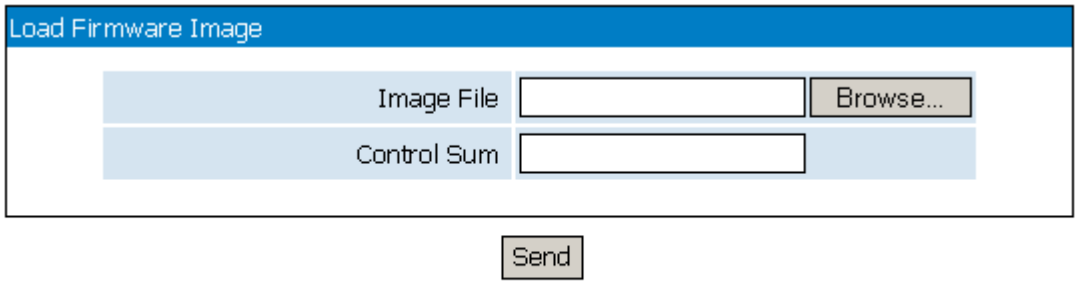
The Upgrade configuration page is simplified, containing just a few elements – The SS-16 is operating in System (Kernel) mode only.



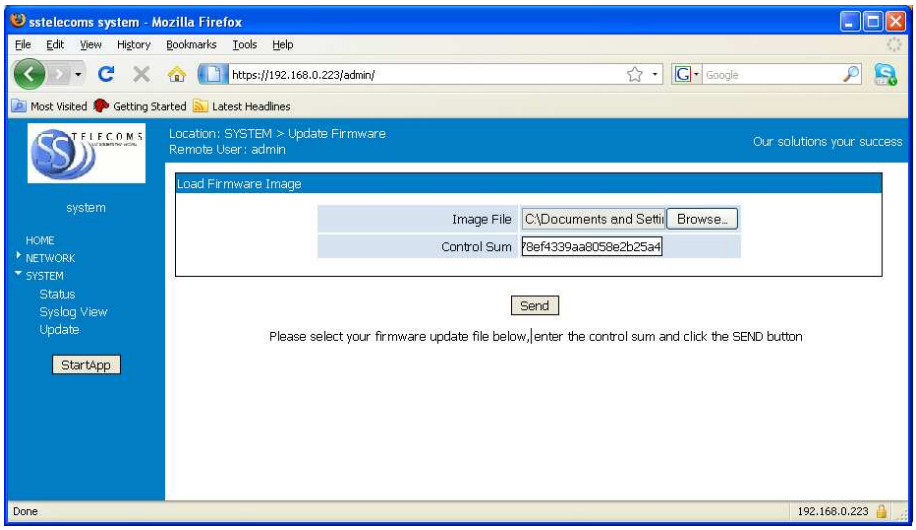
There are only two pages available, NETWORK and SYSTEM. Network shows the state of the network, you can't perform changes. In System, you can see the state of the equipment, or look at the logs, as shown below.



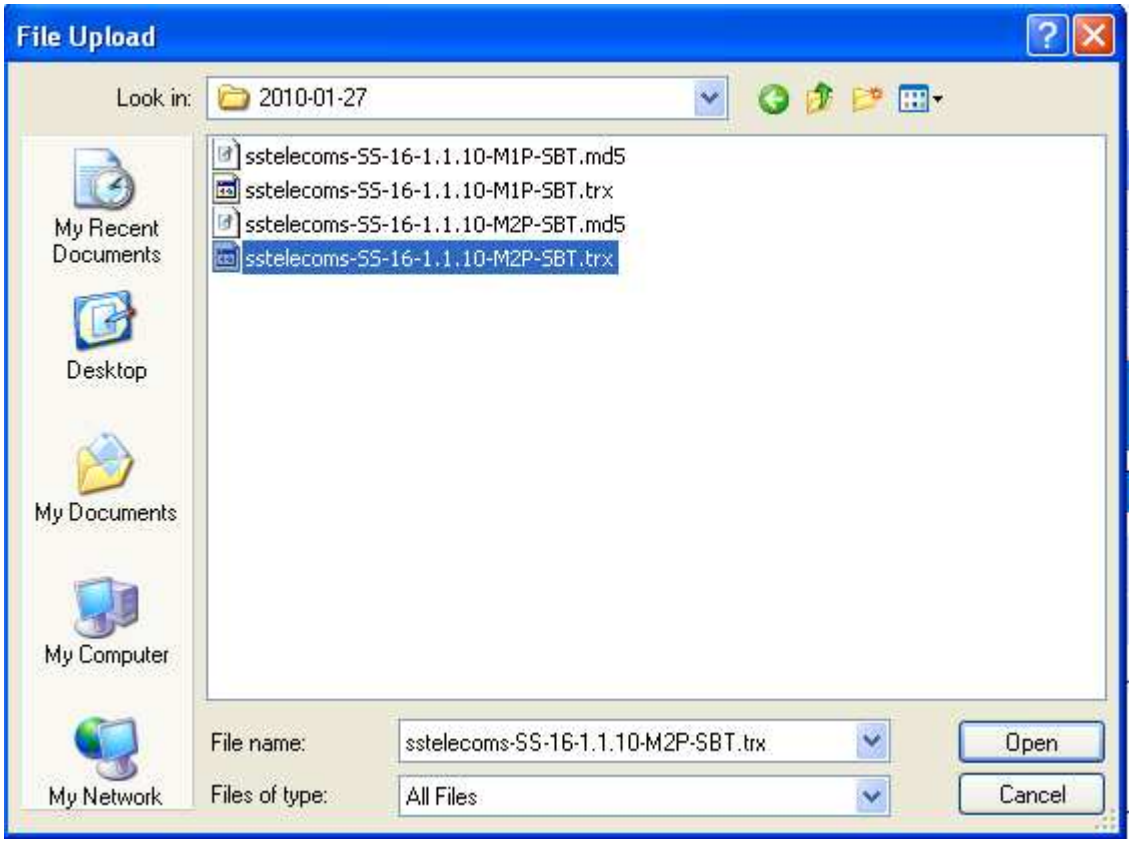
To perform an update, go to SYSTEM>Update as shown in the next image:


Load Firmware Image
Image File
Browse...
Control Sum
Send
Please select your firmware update file below, enter the control sum and click the SEND button

The update (firmware image) must be on the PC. The image files are binaries that have the extension “trx”. One may download the image files from the SS-Telecoms website. The update files are called “sstelecoms-SS-16-1.1.13-M2P-SBT.trx, where “equipment” is the name of the device, such as <<SS-16>> and 1.1.13 Is the version number. The next three letters indicate the hardware model, such as M1P or M2P:

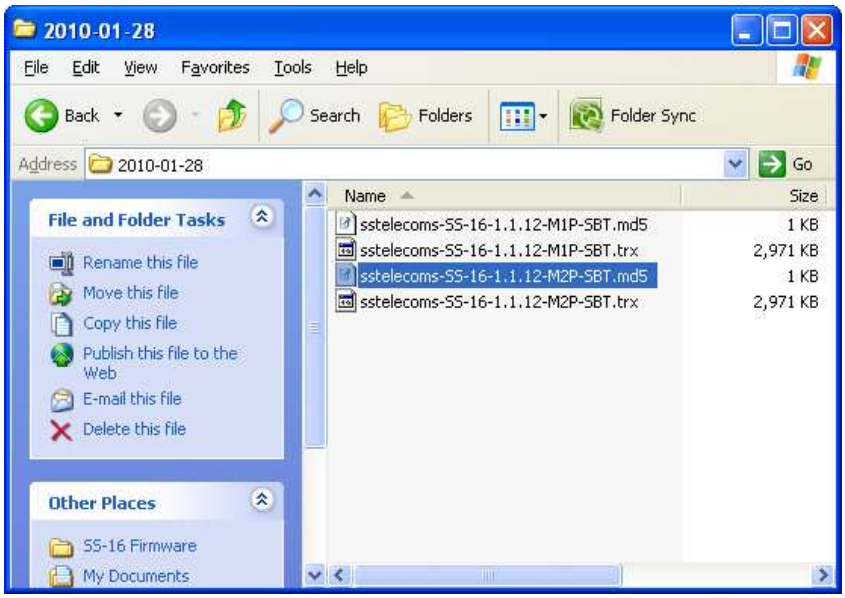


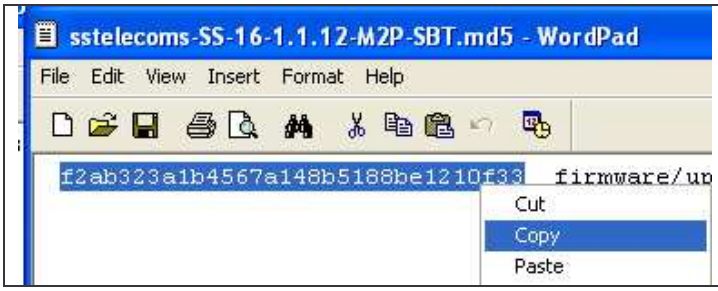
Enter the name of the firmware file or click Browse to search for it on the system. Then select the file needed and click Open.



Note:
*Each firmware file has an associated **Control Sum.** The control sum is a string of hexadecimal figures, such as "f2209c63972be34f55d4e69d90042d93" and it is stored in a text file (with .txt extension), with the same name as the firmware image. Using a control sum prevents the user from loading a corrupted image.*

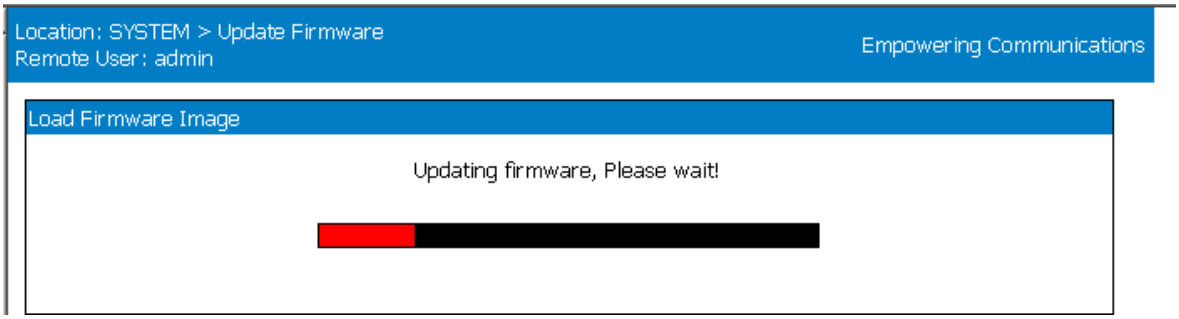
Open the text file with the control sum, as shown below:





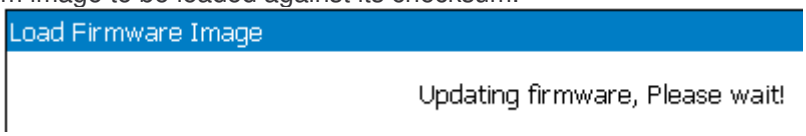
copy the sting of checksum from this file and paste it in the SS-16 IP web-page in the field named "Control Sum, then click the "Send" button to perform the update*:*

A "Firmware Update" window shows up where progress can be seen with a progress indicator under the message "Firmware is updating, please wait!"

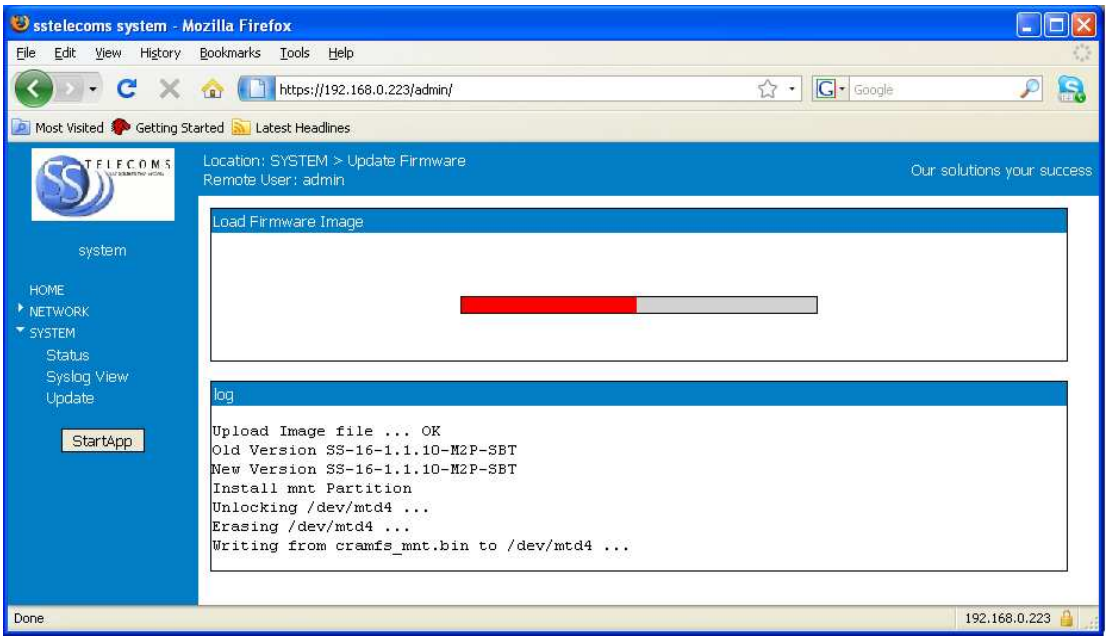


Note that there are **two** distinct, successive phases of updating the firmware image:

- While the message "Updating firmware" is blinking, the equipment just checks the program image to be loaded against its checksum.



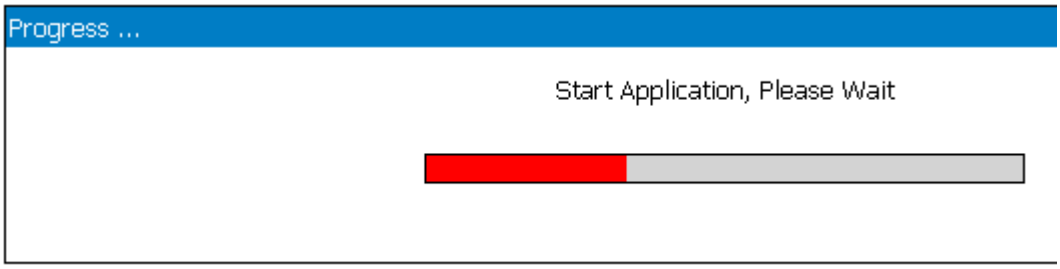
- If the result of the verification is OK, the SS-16 VOIP-2-GSM device goes to the second phase where it loads the new firmware into its Flash memory.



After the loading of the new firmware is completed, you can restart the SS-16 equipment to start it operating with the new software.

For this, you may use the "**StartApp**" button located at the button of the system menu, at the left of the web page.	

The button will start up the main application software running on SS-16



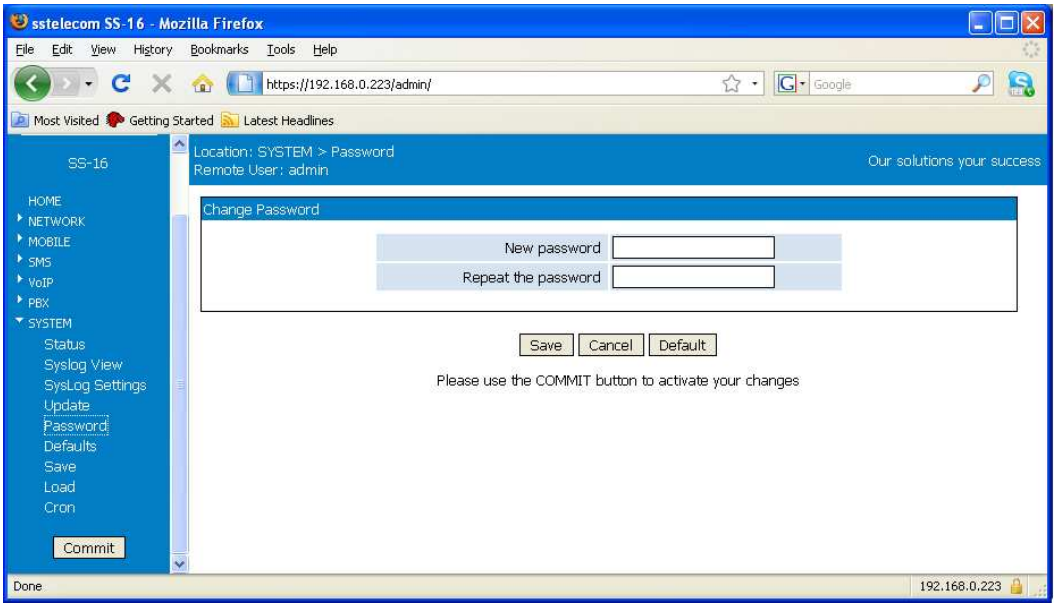
Warning: *Do not update the firmware unless there are problems with the SS-16 VOIP-2-GSM router or the new firmware has a feature that you need. Remember to backup the current configuration first. Be careful when loading an update file. If selecting a wrong file, or if for different reasons firmware upgrading fails, the equipment may no longer operate correctly. A special firmware restore will have to be performed.*

To avoid this, follow the rules carefully:

- don't turn off the SS-16 VOIP-2-GSM equipment or the computer while the firmware is being overwritten;
- remember the equipment does not work while firmware update is in progress;
- after successful updating, verify the upgraded firmware;
- Remember that updating the firmware on the SS-16 VOIP-2-GSM could cause some or all of the configuration settings to be lost, depending on the degree of change in the firmware.
  Therefore it is highly recommended to **save** the current configuration before updating, and then restore it. To backup the settings, perform a Save, update the firmware, and then load the saved settings, after having the equipment operating with the updated firmware.

6.8.5 Password

Allows the modification of passwords for the log-in accounts, in this case "admin". For changing the "admin" account password, the user must be logged in as admin.
Type the new password and enter it again on the second row to confirm it.



"Save" saves the new password.

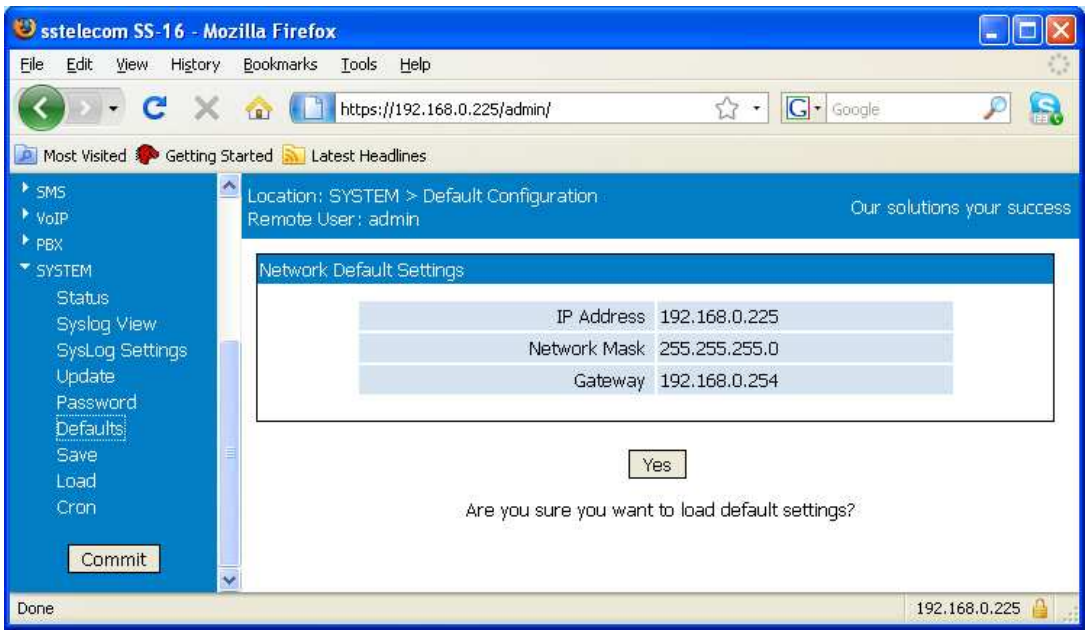
Replace the default password (99admin11) with one of your own as soon as possible.

Please choose a password with a **minimum** of six characters. In order to effectively prevent unauthorized access, the password must be long enough and include both letters and numbers.
Note that the password is **case-sensitive**.

6.8.6 Defaults

This option restores **all** the system settings to factory defaults. One can also click Default on each page to reset the respective parameters to default, but this function restores the whole device to factory default at once.

When selecting "Defaults", a confirmation window shows up, asking you "Are you sure?"



If wanting to proceed, click the YES button and the equipment will revert to the factory default settings.
Following a reset, it will start operating with the factory default values for all parameters.

During the process, “Updating settings please wait!” and a red progress bar, can be seen as shown below:

CAUTION!
Use this option carefully.
During the process, the SS-16 VOIP-2-GSM will not be available for routing and connecting to the 3G network. For a few moments, the LAN icon in the status bar will be barred with a red “x” and the message “A network cable is unplugged” will show up, indicating that SS-16 VOIP-2-GSM is unavailable.

Hardware factory defaults

“Return to Defaults” can also be performed via **hardware**. This means the Web interface doesn’t have to be used.

For this, perform the Hardware reset described previously:
Remember to follow the procedure described next because by simply pressing the recessed RST button a reset will not be performed. Follow these steps:

- shut down the SS-16 VOIP-2-GSM device (unplug the power jack)
- press the RST button and keep it pressed
- power up SS-16 VOIP-2-GSM
- wait until all three LEDs light up in red, and then leave the RST button.

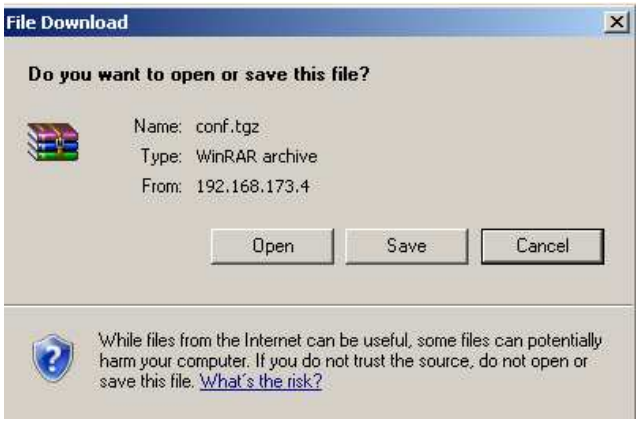
Now the equipment performs a reset with return to factory default settings. This does NOT happed if you don’t follow exactly the procedure described above – for instance, if continuing to press RST after the three indicators have turned red.

This way of restoring the factory default settings is faster than using the web interface, but the same care should be exercised.
Note: T*he “Return to Defaults” option is quite useful when incorrect settings have been performed or when the IP of the SS-16 VOIP-2-GSM device are forgotten. If not having the IP address or if having incorrect settings for iptables, one won’t be able to connect to the VoIP2Mobile to administer it. Perform a hardware “factory defaults”, and SS-16 VOIP-2-GSM will revert to its original settings, including the IP address of 192.168.0.225*

6.8.7 Save

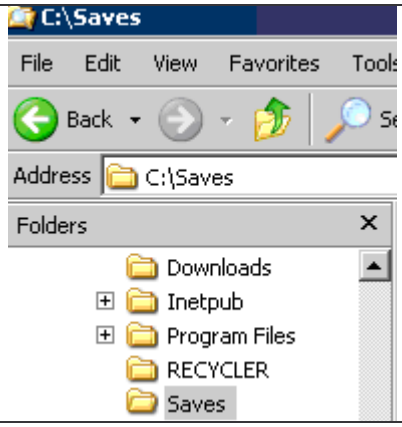
Makes a backup copy of the current configuration (all the settings made) of the SS-16 VOIP-2-GSM.

The configuration file is called “CONF.tar” by default
A message similar to this will be shown:


File Download
Do you want to open or save this file?
Name: conf.tgz
Type: WinRAR archive
From: 192.168.173.4
Open
Save
Cancel
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?

Note: As shown in the example above, the Windows operating system may see this configuration file as a kind of archive and issue a warning, but it does not matter.

Click “Save” or select “Save it to disk” to save the file with the current configuration of the SS-16 VOIP-2-GSM.

One may select the folder where the configuration will be saved; also, one should change the default name “conf.tar” into a name meaningful for the actual configuration.	

Note: *It is highly recommended to backup the configuration from time to time. This way the user can always load the saved configuration, if there is need. For instance, when performing a “Restore to factory defaults”, all previous settings are lost. If the configuration was saved, it can be loaded back to perform changes.*

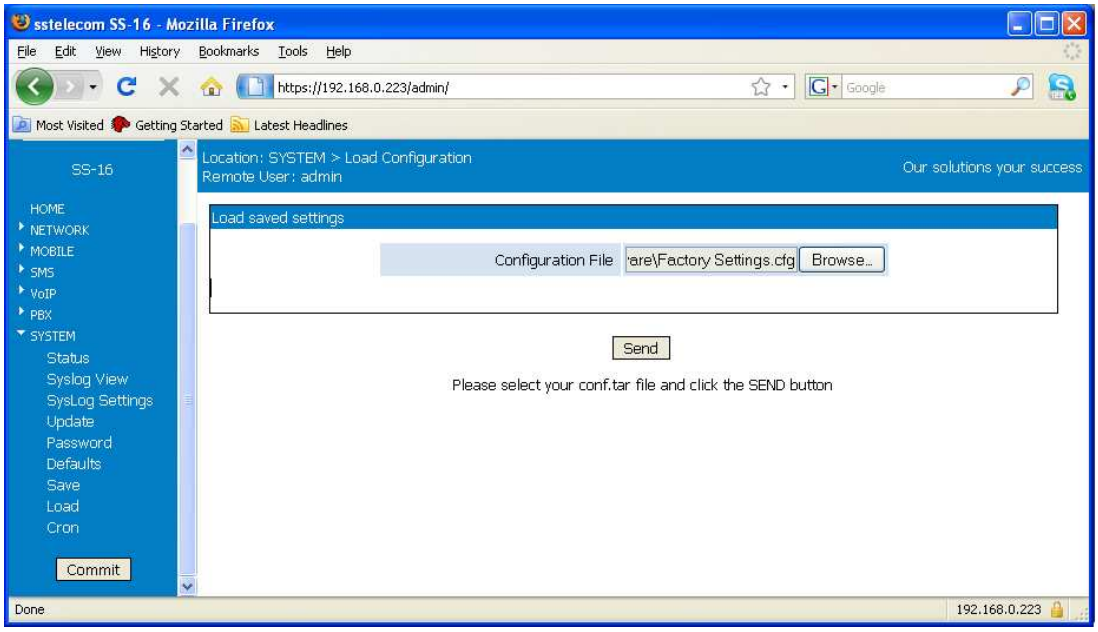
Also, one should perform a “Save” before using “Update” to load a new software image, since the new firmware could erase some of the settings.

This is especially useful when having configured many particular settings, for example related to routing rules or to IP SEC tunnels – it will take a long time to edit these specific settings by hand.

6.8.8 Load

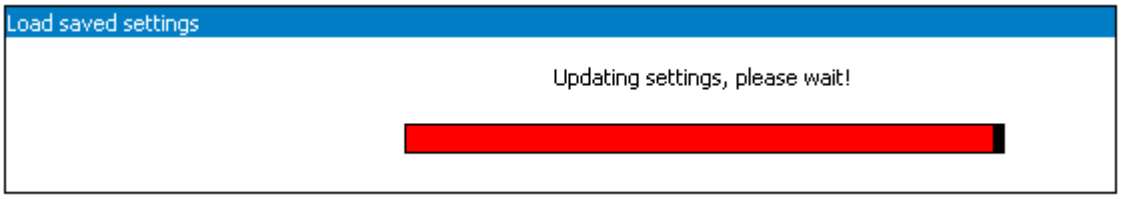
It loads a configuration file that has been saved previously or restores a backup.

It is the operation complimentary to “Save”:



Use “Browse” or “Choose” button to navigate to the location where the configuration file was saved.

Press Open, then click “Send” to send the file to the SS-16 VOIP-2-GSM.



First the message “Updating settings, please wait” will start to blink, and then a red progress, below the blinking message.

7. TECHNICAL SPECIFICATIONS

TITLE	DESCRIPTION
Mobile networks supported	GSM /GPRS and respectively UMTS for 3G modules
Frequency bands	GSM/GPRS/EDGE: Tri-band, 800/900/1800 MHz UMTS 2100MHz: 2100 MHz (Band Class 1)
Supported protocols	TCP/IP, PPP, CHAP, PAP, ARP, UDP, RIP v.2, NTP and others
Features	Web configuration
Wired Network interface	One LAN port, Ethernet 10/100 base-T Connector type: female RJ45 Transfer speed: 10/100 Mbps
Max. transmitter power	GSM/GPRS/EDGE 800/900MHz: +32dBm (Power Class 4) GSM/GPRS/EDGE 1800MHz: +29 dBm (Power Class 1) UMTS/HSDPA 2100MHz: +23dBm (Power Class 3. 8PSK mode Class E2) , only for devices fitted with 3G modules
Receiver static sensitivity	GSM/GPRS/EDGE 800/900: - 107 dBm 1800 MHz: -105 dBm Compliant with TS 05.05 (R99) UMTS/HSDPA 2100MHz: Compliant with 3GPP TS 25.101(R5) , only for devices fitted with 3G modules
Voice service	GSM: FR, EFR, HR, AMR UMTS: AMR(12.2kbps/10.2kbps/7.95kbps/7.4kbps/6.7kbps/5.9kbps /5.15kbps/4.75kbps)
Concurrent service (Based on UMTS)	1 UMTS CS voice + 1 UMTS PS data only for devices fitted with 3G modules
SMS	Supporting SMS based on CS domain of GSM or UMTS Supporting SMS based on PS domain of UMTS
Mobile antennas	External quad-band antennas with magnetic base and cable
Supply voltage	12 $V_{D.C.}$ (+ centre wire, - external conductor)
Supply adapter	Special adapter, Dee Van Enterprises Ltd. Type: DVE DSA-30PF-12 output : 12 $V_{D.C.}$ / 2 A Input: from 100 to 240 $V_{A.C.}$ / 0,3 A / 50 Hz
Status indicators	Three bicolor LEDs, for Status, Mobile 1, Mobile 2
Temp. range	Operating: 5^0 .... 50^0 C Storage: 0^0.... 60^0 C
Dimensions (L x W x H)	222 x 115 x 32 (mm) except protrusions
Weight	500 g – SS-16 VOIP-2-GSM unit 1 kg – the whole package

7.1 OPERATING ENVIRONMENT

SS-16 VOIP-2-GSM was designed for indoor use only, so you should NOT operate it outdoors.

Install the SS-16 VOIP-2-GSM in closed rooms or enclosures, where the environmental conditions should be:

- operating temperature range: from 5 to 50 degrees C
- relative humidity: from 10 to 85 %, non-condensing

Avoid dust and prolonged exposure to sun. In addition, The SS-16 VOIP-2-GSM must NOT be used in a flammable or explosive environment, or in locations where toxic or flammable gases may accumulate.

The equipment must be handled with care, to avoid mechanical shocks and blows.

The SS-16 VOIP-2-GSM should not be used in an environment with high level of EMI (electromagnetic interferences) that is in close proximity to high power equipment such as electric motors or heaters. In addition, it should not be placed near copiers, PC monitors, TV sets or other audio-video appliances.

In order to ensure adequate working temperature, the SS-16 VOIP-2-GSM interface must not be used too close to heat sources or in direct sunlight.
Also, it must not be overcrowded: leave free space around, below and above the SS-16 VOIP-2-GSM router. This is needed both for connecting Ethernet and antennas cables, and also for ventilation (natural air cooling to dissipate the heat generated during operation).

When allowed operating temperature is exceeded this may not have an immediate, visible effect on the SS-16 VOIP-2-GSM unit, but it can result in unreliable operation, accelerated ageing and hence diminishing of lifetime.

When selecting the location for installing the external antennas of the SS-16 VOIP-2-GSM, refer to the recommendations described in the chapter about mounting the equipment.

The SS TELECOMS SS-16 VOIP-2-GSM does not include materials or components that are harmful to the environment.

When the life cycle of this device has expired and it cannot be repaired or re-used, dispose of it in accordance with applicable laws and legal regulations.

8. GLOSSARY

3G – "Third generation" mobile networks, specially designed for high speed data services. The classic definition of wireless networks that following the 2G systems (GSM) and they offer high speed data services in addition to the basic voice capability. These 3G mobile communications systems provide an enhanced range of multimedia services (high speed Internet access, video streaming, etc.). The high data transfer speed specific to the third generation communications network leads to an increased efficiency of information transmission, while the real time access to data and information means important savings of time and money. UMTS is the best known of the 3Gnetworks, while HSDPA is a 3,5G development.
ANI – Acronym for **Automatic Number Identification** - a feature of telephony intelligent network services which allows subscribers to display or capture the telephone numbers of calling parties. The service is often provided by sending the digital tone multi frequency (DTMF) tones along with the call. Home users of ANI can screen callers. ANI is commonly used by emergency centre dispatchers to save the caller having to report the information and, when necessary, to help locate callers. A telephone company's 9-1-1 service to a public safety point usually includes the ANI feature. For instance, in a call center, ANI displays the number of the calling party to the call center agent in real time. Among other things, the call center can use the information to forward calls to different people for different geographic areas.

DHCP (Dynamic Host Configuration Protocol) - This protocol allows a computer (or many computers on your network) to be automatically assigned a single IP address from a DHCP server. DHCP is available on SS-16 VOIP-2-GSM number that the caller dialled. DNIS works by passing the touch tone digits (dual tone multi frequency or MF digits) to the destination where a special facility can read and display them or make them available for call center programming. For example, a company may have a different toll free number for each product line it sells. If a call center is handling calls for multiple product lines, the switch that receives the call can examine the DNIS, and then play the appropriate recorded greeting. Another example of multiple toll free numbers might be used for multi-lingual identification. A dedicated toll free number might be set up for Spanish or Chinese speaking customers.
DNS – Acronym for Domain Name System (or Service), an Internet service that translates domain names into IP addresses. This allows the Internet hosts to use both addresses type domain name (such as SS Telecoms.ro or linux.org), and addresses type IP numbers (for instance 192.17.3.4). The domain name addresses are intended for human users and are automatically converted into IP (numeric) addresses. Because domain names are alphabetic, they are much easier to remember. The Internet however, is really based on IP addresses. Every time you use a domain name, therefore, a DNS service must translate the name into the corresponding IP address.
DNS Server – computer able to answer to the interrogations in a DNS system. The DNS server maintains a database that includes the host computers with their domain names and the corresponding IP addresses. For instance, if you ask the DNS server for the domain name apex.com, it will return the IP address of the hypothetical company called Apex. DNS servers are linked in their network, so if one DNS server doesn't know how to translate a particular domain name, it asks another one, and so on, until the correct IP address is found. When a user enters a domain name into the Internet browser, the user is sent by the DNS Server to the proper IP address. The DNS server address used by the computers on your home network is the location of the DNS
DNS Server Address (Domain Name System) - DNS allows Internet host computers to have a domain name and one or more IP addresses. A DNS server keeps a database of host computers and their respective domain names and IP addresses, so that server your ISP has assigned.

Dynamic DNS - service that allows clients connecting to the Internet with a dynamic IP address to be able to use applications that require a static IP address. The Internet Service Provider changes the IP address of the users, but there are applications that work only with static (fixed) IP addresses. Dynamic DNS makes it possible for sites on the Internet to establish connections to you computer without need for tracking the IP address themselves. DDNS is useful both for wired or wireless (such as 3G) dialup connection where at each connection a new address is assigned, and for DSL services where the address is changed occasionally by the ISP.
Domain Name Resolution - The resolving of a domain name. Internet applications don't communicate with domain names such as google.com or SS Telecoms.ro, instead they use IP addresses (for example 193.226.61.95 or 216.239.39.99). Domain Name Resolution is the process (transparent for the user) of converting domain names into corresponding IP addresses. Every operating system has routines that deal with resolution of domain names.
Encryption - This provides wireless data transmissions with a level of security.
Ethernet - Standard for wired computer networks. Ethernet networks are connected by cables and hubs, and move data around. For wired connections, SS-16 VOIP-2-GSM provides one Ethernet 10/100 port, for connection to the local LAN.
Firewall - System designed to prevent unauthorized access to or from a private network. The firewall determines which information passes in and out of and prevents anyone outside of your network from accessing your computer without authorization and possibly damaging or viewing your files. Any company with an intranet that allows its workers access to the wider Internet must use a software or hardware firewall to prevent outsiders from accessing its own private data resources and for controlling what outside resources its own users have access to. Here the firewall is software a set of related programs (residing on the gateway server) that protect the resources of the local (internal) network.
Gateway - A network point that manages all the data traffic of your network, as well as to the Internet and connects one network to another. SS-16 VOIP-2-GSM is also gateway, since it interfaces between VoIP calls form the local network and the voice mobile networks.
IP Address – Short from Internet Protocol address. The numerical address of a network device or resource as expressed in the format specified in the Internet Protocol (IP). In the current addressing format, IP version 4, the IP address is a 32-bit (4 bytes) sequence divided into four groups of decimal numbers separated by periods ("dots"). Each number can be zero to 255. These four groups of numbers look like "127.0.0.1" or „213.154.120.170". The four numbers in an IP address are used in different ways to identify a particular network and a host on that network.
IP – Acronym for Internet Protocol. It is the protocol from TCP/IP that directs the way data is sent from one computer to another on the Internet. The messages are divided into data packets, routed from the sender network to the receiver network and there re-assembled in the right order to re-create the original message. Because a message is divided into a number of packets, each packet can, if necessary, be sent by a different route across the Internet. These data packets can arrive in a different order than the order they were sent in. The Internet Protocol just delivers them. It's up to TCP protocol (Transmission Control Protocol) to put the packets back in the right order.
Note that IP is a connectionless protocol, which means that there is no continuing connection between the end points that are communicating. IP corresponds to the layer 3, the Networking Layer from the Open Systems Interconnection (OSI) communication model.
IP Tunneling - a tunnelling protocol encapsulates a packet of the same or lower protocol, while in a regular protocol, the lower layer protocol encapsulates the higher level protocol. In order to achieve a Virtual Private Network (corporate network) you must ensure the port forwarding (tunnelling), that is the transmission of private data through a public network such as 3G. The routing nodes in the respective public network must not be aware that the transmission is part of a private network. Tunneling means the encapsulation of the data and protocol information of the private network within the transmission units of the public network. Widely used tunnelling

methods are the Point-to-Point Tunneling Protocol (PPTP) developed by Microsoft and generic routing encapsulation (GRE), developed by Cisco Systems.
ISP (Internet Service Provider) - An ISP is a business that allows individuals or businesses to connect to the Internet. Users log on to the Internet using an account with an ISP or Internet Service Provider. ISPs can serve IP addresses dynamically, or assign static (fixed) IP addresses to individual computers. In this case, the Internet provider is the operator of the 3G mobile network.
LAN - Acronym for local area network (computer network that spans a relatively small area). A group of computers, workstations and associated devices that share a common communications line or wireless link and are located in a relatively limited area, typically inside the same office building. The communications link that interconnects these computers allows any device of the network to interact with any other from the same network. The devices that compose a local network (workstations, personal computers and peripherals) are called nodes and typically share the resources of a single processor or server. This server has resources (applications, processing and data storage capabilities) that are shared in common by multiple computer users. The LAN can be connected to other local networks over any distance via phone lines, wireless links or other connections, and the system of LANs connected in this way is called a wide-area network (WAN). The LAN may also be connected to the Internet through a gateway. SS-16 VOIP-2-GSM equipment is connected in a local network through its Ethernet port.

LCR – Acronym for **Least Cost Routing** - Refers to the method for delivering calls according to the lowest cost carrier. Often more than one carrier can compete to deliver a call and depending on their relative charges, the LCR Switch (which contains a database of our carriers, destinations and charges) will connect the call via the cheapest carrier. This process means searching for the optimum traffic routes via one or several traffic routings enabling a cost reduction. SS-16 VOIP-2-GSM implemnts LCR functions, since it analyse incoming calls and sends them to the route or towards the GSM module which assures the minimum costs.

MAC Address (Media Access Control) - A MAC address is the hardware (physical) address of a device connected to a network.
All network interface controllers must have a hard-coded and unique MAC address. The MAC address is applied at the factory and uniquely identifies network hardware, such as an Ethernet cards, modems or wireless adaptors on a LAN or WAN. The first part of the address is unique to the company that produced the device, and beyond that it is a sequence of digits unique to a single device manufactured by a company. SS-16 IP features a single Ethernet port, so it will have one MAC.
SNTP - Acronym for **S**imple **N**etwork **T**ime **P**rotocol. Simplified version of NTP, Internet standard protocol used to update the real-time clock in a computer. It assures accurate synchronization to the millisecond of computer clock times in a network of computers.NTP is very useful in packet-switched, variable-latency data networks. In case of SS-16 VOIP-2-GSM, if you want to measure the performances of the network, you need accurate, universal time-stamps for the data packets.
NTP runs as a continuous background client program on a computer and it sends periodic time requests to servers, obtaining server time stamps and using them to adjust the client's clock. NTP uses UDP port 123 as its transport layer. It is designed particularly to resist the effects of variable latency. There are numerous primary and secondary servers in the Internet that are synchronized to the Coordinated Universal Time (UTC) via radio, satellite or modem. For more information, visit www.ntp.org.
Routing -The process of determining and prescribing the path or method to be used for establishing connections and forwarding data packets. In a network, a 'routing switch' is a device that combines the functions of a switch, which forwards data by looking at a physical device address, and a router, which forwards packets by locating a next hop address. Each host with a router in the network uses the routing table information to determine the next host to route a packet to for a specified destination. SS-16 VOIP-2-GSM performs as a router for the wired or wireless local network where it is connected. It can perform both static routing (with fixed, pre-defined routes) and dynamic routing. Dynamic routing is more complex than static routing, but it provides several benefits. It ensures scalability and adaptability. The routes are

dynamic, interactive, permanently updated. Routers learn about the network topology by communicating with other routers so it may select a better route, optimized for each time period.

S-HTTP - Acronym for SECURE HYPERTEXT TRANSFER PROTOCOL. A secure way of transferring information over the Web, by using an application-level encryption scheme. S-HTTP is an extension of the normal HTTP with security enhancements for WWW-based commerce. Web pages that use S-HTTP have a URL starting with https://. Typically HTTP data is sent over TCP/IP port 80, but HTTPS data is sent over port 443. The standard was developed for secure transactions, and uses 40-bit encryption ("weak" encryption) or 128-bit ("strong" encryption). The HTTPS standard supports certificates and allows encryption, digital signatures, authentication, or any combination of these. The S-HTTP type of transaction security is more secure than a user ID and password, thus it is mostly used by financial institutions (for example for credit-card purchases over the Web). Since SS-16 VOIP-2-GSM provides ensures S-HTTP, this means a higher degree of security for you.

SMS - **Acronym** for **Short Message Service**; means the transmission of short text messages to and from cellular phones. The messages must be text only (no images or graphics) and not longer than 160 alpha-numeric characters. Operators of Mobile Phone Networks use a spare data channel to send SMS messages. You may send SMS messages to another mobile subscriber, the mobile operator can send you phone settings over-the-air or commercial companies may send dedicated content to your mobile terminal. The embedded firmware allows SS-16 VOIP-2-GSM to send and receive SMS (via mobile network) from the computer connected to it.

Static packet filter - One of the simplest and least expensive forms of firewall protection is known as static packet filtering. With static packet filtering, each packet entering or leaving the network is checked and either passed or rejected depending on a set of user-defined rules. Dealing with each individual packet, the firewall applies its rule set to determine which packet to allow or disallow. The static packet filtering firewall examines each packet based on the following criteria:

* Source IP address
* Destination IP address
* TCP/UDP source port
* TCP/UDP destination port

Static packet filtering is easy to implement and configure and does not use a lot of resources, but its efficiency is limited. Basic packet filtering firewalls are susceptible to IP spoofing, where an intruder tries to gain unauthorized access to computers by sending messages to a computer with an IP address indicating that the message is coming from a trusted host. Another shortcoming is that this form of firewall rarely provides sufficient logging or reporting capabilities.

SIP – Acronym for **Session Initiation Protocol** - a protocol proposed standard for initiating, modifying, and terminating an interactive user session that involves multimedia elements such as video, voice, instant messaging, online games, and virtual reality. It is one of the leading signalling protocols for Voice over IP, along with H.323. SS-16 VOIP-2-GSM is used for VoIP calls that use the SIP protocol.

SIP clients traditionally use TCP and UDP port 5060 to connect to SIP servers and other SIP endpoints. SIP is primarily used in setting up and tearing down voice or video calls. However, it can be used in any application where session initiation is a requirement. These include Event Subscription and Notification, Terminal mobility and so on. There are a large number of SIP-related RFCs that define behaviour for such applications. All voice/video communications are done over separate transport protocols, typically RTP.

TCP/IP – Acronym for Transmission Control Protocol / Internet Protocol. It was established by the Defense Department of the USA for communications between computers. It has been at first incorporated in Unix operating system but has become the de facto standard for data transmission via networks, including for Internet. Currently TCP/IP is the basic communication language or protocol of the Internet, but it can also be used as a communications protocol in private network. TCP/IP uses the client/server model of communication in which a computer user (a client) requests and is provided a service (such as sending a Web page) by another computer (a server) in the network. TCP/IP is a two-layer program. The higher layer, Transmission

Control Protocol, manages the assembling of a message or file into smaller packets that are transmitted over the Internet and received by a TCP layer that reassembles the packets into the original message. The lower layer, Internet Protocol, handles the address part of each packet so that it gets to the right destination. For details see the definitions of those protocols that Access -LAN supports.
TCP – Acronym for Transmission Control Protocol. In TCP/IP, the TCP part is the one that takes care of keeping track of the individual units of data (packets) that a message was divided into for efficient routing through the Internet. TCP is known as a connection-oriented protocol, which means that a connection is established and maintained until the message or messages to be exchanged by the application programs at each end have been exchanged. TCP is responsible for ensuring that a message is divided into the packets that IP manages and for reassembling the packets back into the complete message at the other end.
At the destination, TCP reassembles the individual packets and waits until they have all arrived to forward them to you as a single file. It also checks the received packets. TCP acts at the transport level (level 4) of the ISO/OSI model. See also ISO/OSI model, packet, TCP/IP.

TDM – Acronym for **Time Division Multiplexing** - is a type of digital multiplexing in which two or more apparently simultaneous channels are derived from a given frequency spectrum by interleaving pulses representing bits from different channels. In some TDM systems, successive pulses represent bits from successive channels. In other systems different channels take turns using the channels for a group of successive pulse-times (a so-called "time slot"), such as voice channels in E1/T1 landline systems or wireless GSM systems. What distinguishes coarse time-division multiplexing from packet switching is that the time-slots are pre-allocated to the channels, rather than arbitrated on a per-time slot basis. Since it interfaces between VoIP and GSM networks, SS-16 VOIP-2-GSM also performs conversion between TDM and packet-switching!

UDP – Acronym for User Datagram Protocol. It is a simpler protocol than TCP/IP that corresponds to the transport layer of the ISO/OSI model. UDP converts the messages generated by the application into data packets to be sent through IP, but does not check if the messages have been transmitted correctly or not.
UDP allows individual packets to be dropped (with no retries) and UDP packets to be received in a different order than they were sent. Consequently UDP is more efficient but less reliable than TCP and is used to different purposes - primarily for broadcasting messages over a network. With UDP, reliability is wholly in charge of the application that generates the message.
UDP is used often in applications such as videoconferencing or games where optimal performance is preferred over guaranteed message delivery.

VoIP - Acronym for **Voice over IP**. It describes the hardware or software category which allows people to make phone calls over the Internet. The voice signals are converted in data packets and these are sent on the public telephonic lines, avoiding the costs of the PSTN network. The VoIP applications can be used with a regular microphone and computer speakers, but IP phones or VoIP speakers can also can be used, providing an identical experience like the regular telephony. The quality and the reliability of the VoIP technology improved so much that many users completely renounced the standard telephony contracts for VoIP technology.

The manufacturer reserves the right to modify the product and manual for the purpose of technical improvement without prior notice. The manufacturer guarantees the good functioning of the product provided that it has been correctly installed and the directives for storage and usage have been respected. The warranty implies exclusively repairing or replacing the defective unit. The warranty does not include any indirect losses or loss of profit. The manufacturer is not liable for any damage, whether direct, indirect, special, incidental, or consequential, as a result of using SS Telecoms SS-16 VOIP-2-GSM .

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It is certified hereby that the SS Telecoms SS-16 VOIP-2-GSM unit is manufactured in concordance with the legal provisions concerning responsibility towards the quality of delivered products, fulfills the quality parameters specified in its "User's manual" and is fit for the purpose for which it has been designed. It also warrants that the equipment will perform substantially in accordance with the accompanying documentation.

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